

---

# MultiVOIP™ FX SIP Gateways

User Guide for Voice/IP Gateways

Models: MVPFX2-2  
MVPFX2-4  
MVPFX2-8



---

## User Guide

### S000411A

MultiVOIP FX Analog SIP Gateways (Models MVPFX2-2, MVPFX2-4 & MVPFX2-8)

This publication may not be reproduced, in whole or in part, without prior expressed written permission from Multi-Tech Systems, Inc. All rights reserved.

Copyright © 2006, by Multi-Tech Systems, Inc.

Multi-Tech Systems, Inc. makes no representations or warranties with respect to the contents hereof and specifically disclaims any implied warranties of merchantability or fitness for any particular purpose. Furthermore, Multi-Tech Systems, Inc. reserves the right to revise this publication and to make changes from time to time in the content hereof without obligation of Multi-Tech Systems, Inc. to notify any person or organization of such revisions or changes.

### Record of Revisions

Revision	Description
A	06/29/06. Initial release. Describes 13.01 software release.

### Patents

This Product is covered by one or more of the following U.S. Patent Numbers: *6151333, 5757801, 5682386, 5.301.274; 5.309.562; 5.355.365; 5.355.653; 5.452.289; 5.453.986.* Other Patents Pending.

### Trademark

Trademark of Multi-Tech Systems, Inc. is the Multi-Tech logo. Windows is a registered trademarks of Microsoft.

GENERAL CONTACT	TECHNICAL SUPPORT		
Multi-Tech Systems, Inc. 2205 Woodale Drive Mounds View, Minnesota 55112, USA (763) 785-3500 (800) 328-9717 Fax: 763-785-9874 <a href="http://www.multitech.com">www.multitech.com</a>	Country	By E-mail	By Phone
	U.S. & Canada	<a href="mailto:tsupport@multitech.com">tsupport@multitech.com</a>	(800) 972-2439
	France	<a href="mailto:support@multitech.fr">support@multitech.fr</a>	(33) 1-64 61 09 81
	India	<a href="mailto:support@multitechindia.com">support@multitechindia.com</a>	(91) 124-340778
	U.K.	<a href="mailto:support@multitech.co.uk">support@multitech.co.uk</a>	(44) 118 959 7774
Rest of World	<a href="mailto:support@multitech.com">support@multitech.com</a>	(763) 785-3500	

---

# CONTENTS

<b>CHAPTER 1: OVERVIEW</b> .....	<b>5</b>
ABOUT THIS MANUAL .....	6
INTRODUCTION TO ANALOG MULTIVOIP FX SIP GATEWAY VOICE-OVER-IP UNITS (MVPFX2-2/4/8) .....	7
<i>MultiVOIP Front Panel LEDs</i> .....	10
COMPUTER REQUIREMENTS .....	12
SPECIFICATIONS .....	12
INSTALLATION AT A GLANCE .....	13
RELATED DOCUMENTATION .....	13
<b>CHAPTER 2: QUICK START GUIDE</b> .....	<b>14</b>
MULTIVOIP STARTUP TASKS .....	15
<i>Phone/IP Details *Absolutely Needed* Before Starting the Installation</i> .....	16
Gather IP Information .....	16
Gather Telephone Information (Analog) .....	16
Config Info CheckList .....	17
Identify Remote VOIP Site to Call .....	17
<i>Command/Control Computer Setup (Specs &amp; Settings)</i> .....	18
<i>Placement</i> .....	18
<i>Quick Hookup for MVPFX2-4 &amp; MVPFX2-8</i> .....	19
<i>Quick Hookup for MVPFX2-2</i> .....	20
<i>Ensure that Java &amp; Browser Versions will Support Web-Based GUI</i> .....	21
<i>Changing the IP Address through the Console Connection</i> .....	21
<i>Phone/IP Starter Configuration</i> .....	28
<i>Phonebook Starter Configuration (with remote voip)</i> .....	29
Outbound Phonebook .....	29
Inbound Phonebook .....	32
<i>Phonebook Tips</i> .....	34
<i>Phonebook Example</i> .....	36
<i>Connectivity Test</i> .....	41
<i>Troubleshooting</i> .....	43
<b>CHAPTER 3: MECHANICAL INSTALLATION AND CABLING</b> .....	<b>44</b>
INTRODUCTION .....	45
SAFETY WARNINGS .....	45
<i>General Safety</i> .....	45
<i>Lithium Battery Caution</i> .....	45
<i>Ethernet (WAN) Ports Caution</i> .....	45
<i>Safety Warnings Telecom</i> .....	46
UNPACKING YOUR MULTIVOIP .....	47
<i>Unpacking the MVPFX2-4/8</i> .....	47
<i>Unpacking the MVPFX2-2</i> .....	48
<i>Safety Recommendations for Rack Installations of MVPFX2-4 or MVPFX2-8</i> .....	50
<i>19-Inch Rack Enclosure Mounting Procedure</i> .....	51
CABLING PROCEDURE FOR MVPFX2-4/8 .....	52
<i>Cabling Procedure for MVPFX2-2</i> .....	53
<b>CHAPTER 4: MULTIVOIP &amp; AUXILIARY SOFTWARE</b> .....	<b>55</b>
INTRODUCTION .....	56
SUMMARY .....	56
<b>CHAPTER 5: TECHNICAL CONFIGURATION</b> .....	<b>57</b>
CONFIGURING THE MULTIVOIP .....	58
CONFIGURATION BY WEB GUI .....	59
<i>Pre-Requisites</i> .....	59
IP Parameters .....	59
Telephony Interface Parameters .....	60
Config Info CheckList .....	60

<i>Procedure for Configuration by Web GUI (Summary)</i> .....	61
<i>Local Configuration Procedure (Detailed)</i> .....	61
<b>CHAPTER 6: PHONEBOOK CONFIGURATION</b> .....	<b>101</b>
CONFIGURING MULTIVOIP PHONEBOOKS .....	102
PHONEBOOK EXAMPLES .....	113
<i>2 Site Example</i> .....	113
<i>Configuring Mixed Digital/Analog VOIP Systems</i> .....	119
<i>Call Completion Summaries</i> .....	124
<i>Variations in PBX Characteristics</i> .....	127
<b>CHAPTER 7: OPERATION AND MAINTENANCE</b> .....	<b>128</b>
OPERATION AND MAINTENANCE SUMMARY .....	129
<i>System Information screen</i> .....	130
<i>Statistics Screens</i> .....	133
<i>About Call Progress</i> .....	133
<i>About IP Statistics</i> .....	138
GENERAL OPERATION FUNCTIONS .....	142
<i>Change Username/Password</i> .....	142
Establishing a Username and Password.....	142
About Passwords & Login/Logout from Specific Computers .....	143
<i>Logout</i> .....	144
<i>Save &amp; Apply</i> .....	144
<i>Reboot Voip</i> .....	145
<i>Restore Factory Defaults</i> .....	145
UPGRADING MULTIVOIP FIRMWARE.....	146
<i>Introduction</i> .....	146
<i>Identifying Current Firmware Version</i> .....	146
<i>Obtaining Updated Firmware</i> .....	147
UPGRADING MULTIVOIP FIRMWARE VIA FTP CLIENT AND VOIP'S BUILT-IN FTP SERVER FUNCTION.....	149
SYSLOG SERVER FUNCTIONS .....	171
<b>CHAPTER 8 WARRANTY, SERVICE, AND TECH SUPPORT</b> .....	<b>174</b>
LIMITED WARRANTY.....	175
REPAIR PROCEDURES FOR U.S. AND CANADIAN CUSTOMERS .....	175
TECHNICAL SUPPORT .....	176
<i>Contacting Technical Support</i> .....	176
<b>CHAPTER 9: REGULATORY INFORMATION</b> .....	<b>177</b>
<i>EMC, Safety, and R&amp;TTE Directive Compliance</i> .....	178
FCC DECLARATION.....	178
<i>Industry Canada</i> .....	178
<i>FCC Part 68 Telecom</i> .....	178
<i>Canadian Limitations Notice</i> .....	179
<i>WEEE Statement</i> .....	180
<b>APPENDIX A: CABLE PINOUTS</b> .....	<b>181</b>
APPENDIX A: CABLE PINOUTS.....	182
<i>Command Cable</i> .....	182
<i>Ethernet Connector</i> .....	182
<b>APPENDIX B: TCP/UDP PORT ASSIGNMENTS</b> .....	<b>183</b>
WELL KNOWN PORT NUMBERS .....	184
PORT NUMBER ASSIGNMENT LIST.....	184
<b>INDEX</b> .....	<b>185</b>

---

# Chapter 1: Overview

## About This Manual

This manual is about Voice-over-IP products made by Multi-Tech Systems, Inc. It describes three analog MultiVOIP™ FX units that operate with the SIP transmission protocol only, namely, models MVPFX2-2, MVPFX2-4, and MVPFX2-8.

These MultiVOIP units can inter-operate with other contemporary analog MultiVOIP units (MVP130, MVP130FXS, MVP210, MVP410, and MVP810), with contemporary SIP-Survivability MultiVOIP units (MVP210-SS, MVP410-SS, and MVP810-SS), with contemporary BRI MultiVOIP units (MVP410ST & MVP810ST), with contemporary digital T1/E1/ISDN-PRI MultiVOIP units (MVP2410 and MVP3010), and with the earlier generation of MultiVOIP products (MVP200, MVP400, MVP800, MVP120, etc.)

The table below describes the vital characteristics of the various models described in this manual.

Analog MultiVOIP SIP-Only Gateways			
Description-Model	MVPFX2-8	MVPFX2-4	MVPFX2-2
<b>Function</b>	analog voip gateway, SIP only, web GUI only (no Windows GUI)	analog voip gateway, SIP only, web GUI only (no Windows GUI)	analog voip gateway, SIP only, web GUI only (no Windows GUI)
<b>Capacity</b>	8 channels	4 channels	2 channels
<b>Chassis/Mounting</b>	19" 1U rack mount	19" 1U rack mount	table-top unit

**How to Use This Manual.** *In short, use the index and the examples.*

When our readers crack open this large manual, they generally need one of two things: information on a very specific software setting or technical parameter (about telephony or IP) *or* they need help when setting up phonebooks for their voip systems. The index gives quick access to voip settings and parameters. It's detailed. Use it. The best way to learn about phonebooks is to wade through examples like those in our Phonebook Configuration chapter. Finally, this manual is meant to be comprehensive. If you notice that something important is lacking, please let us know.

**Additional Resources.** The MultiTech web site ([www.multitech.com](http://www.multitech.com)) offers both a list of Frequently Asked Questions (the MultiVOIP FAQ) and a collection of resolutions of issues that MultiVOIP users have encountered (these are Troubleshooting Resolutions in the searchable Knowledge Base).

## Introduction to Analog MultiVOIP FX SIP Gateway Voice-over-IP Units (MVPFX2-2/4/8)

**VOIP: The Free Ride.** We proudly present Multi-Tech's MVPFX2-2/4/8 MultiVOIP™ FX SIP Gateways. These three models allow voice/fax communication to be transmitted at no additional expense over your existing IP network, which has ordinarily been data only. To access this free voice and fax communication, you simply connect the MultiVOIP to your telephone equipment and your existing Internet connection. These analog MultiVOIPs inter-operate readily with T1 or E1 MultiVOIP units.



Figure 1-1: MVPFX2-8 Chassis

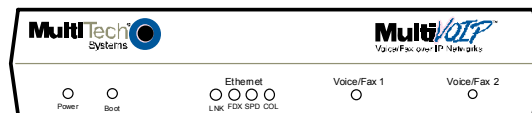


Figure 1-2: MVPFX2-2 Chassis

**Capacity.** The MultiVOIP FX model MVPFX2-8 is an eight-channel unit, the model MVPFX2-4 is a four-channel unit, and the MVPFX2-2 is a two-channel unit. All three of these MultiVOIP units have a 10/100Mbps Ethernet interface for its full-featured web-based configuration GUI and a console port for local access to basic startup configuration parameters (like the gateway's IP address and password).

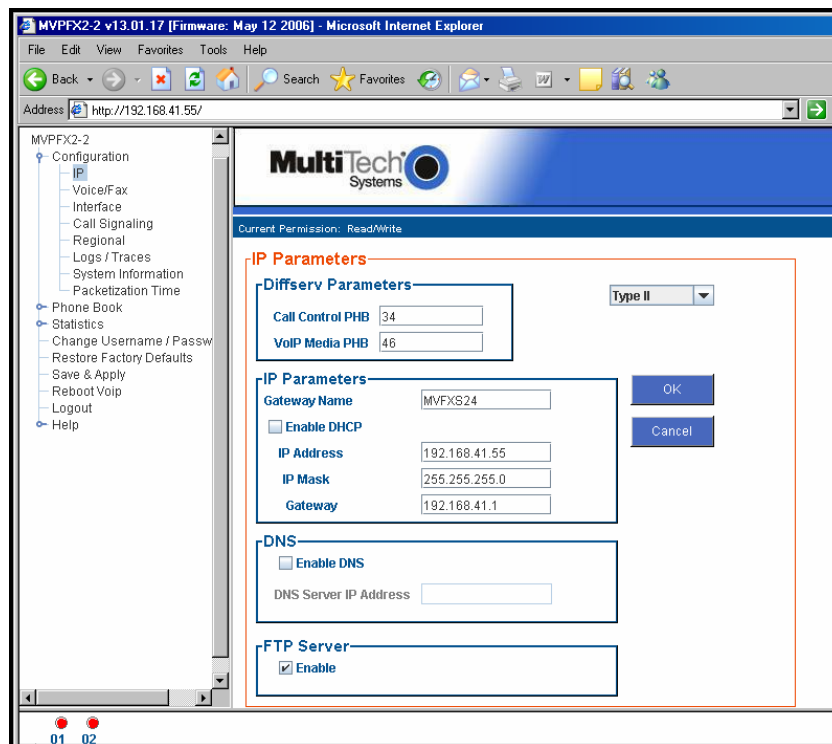
**Mounting.** Mechanically, the MVPFX2-4 and MVPFX2-8 MultiVOIP FX units are designed for a one-high industry-standard EIA 19-inch rack enclosure. The product must be installed by qualified service personnel in a restricted-access area, in accordance with Articles 110-16, 10-17, and 110-18 of the National Electrical Code, ANSI/NFPA 70. The MVPFX2-2 is a table-top unit.

**Phone System Transparency.** These MultiVOIPs inter-operate with a telephone switch or PBX, acting as a switching device that directs voice and fax calls over an IP network. The MultiVOIPs have “phonebooks,” directories that determine to who calls may be made and the sequences that must be used to complete calls through the MultiVOIP. The phonebooks allow the phone user to interact with the VOIP system just as they would with an ordinary PBX or telco switch. When the phonebooks are set, special dialing sequences are minimized or eliminated altogether. Once the call destination is determined, the phonebook settings determine whether the destination VOIP unit must strip off or add dialing digits to make the call appear at its destination to be a local call.

**Voip Protocol.** The MVPFX units use the SIP protocol only. (“SIP” means Session Initiation Protocol.)

**Data Compression & Quality of Service.** The analog MultiVOIP™FX unit comes equipped with a variety of data compression capabilities, including G.723, G.729, and G.711 and features DiffServ quality-of-service (QoS) capabilities.

**Management.** Configuration and system management for the MVPFX2 units is done primarily through a web interface. Once you know the IP address of an MVPFX2 unit, you can contact that unit with a web browser and set the unit’s operating parameters, which are grouped into several separate screens.



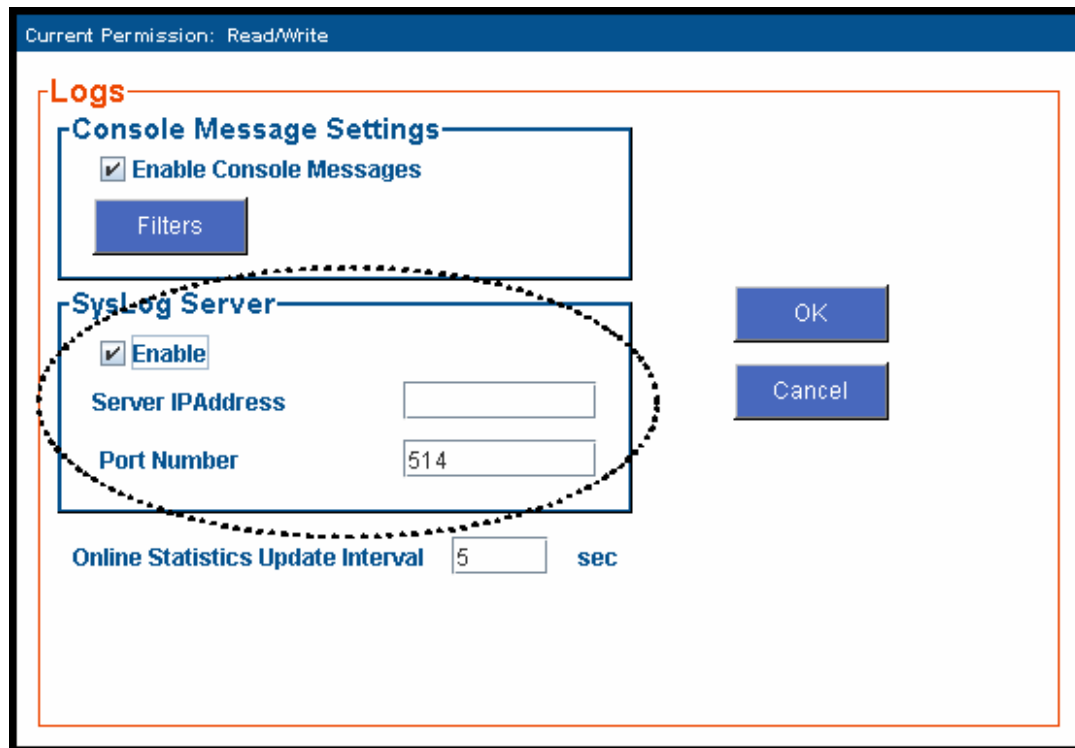
**Figure 1-3: The Presentation of the MultiVOIP Web-Based GUI (IP Parameters screen shown)**

Certain base-level parameters (like the IP address and password of the unit) can be set by connecting the MVPFX2 unit’s “Console” receptacle to a serial connector on a PC (using a DB9-to-RJ45 connector).

The primary advantage of the web GUI is remote access for control and configuration. The controller PC and the MultiVOIP unit itself must both be connected to the same IP network and their IP addresses must be known.



**Logging of System Events.** MultiTech has built SysLog Server functionality into the software of the MultiVOIP units. SysLog is a *de facto* standard for logging events in network communication systems.



The screenshot shows a configuration window titled "Current Permission: Read/Write". The window is divided into two main sections: "Console Message Settings" and "SysLog Server".

- Console Message Settings:** Includes a checked checkbox for "Enable Console Messages" and a "Filters" button.
- SysLog Server:** Includes a checked checkbox for "Enable", a "Server IPAddress" text input field, a "Port Number" text input field containing the value "514", and an "Online Statistics Update Interval" text input field containing the value "5" followed by the unit "sec".

On the right side of the window, there are two buttons: "OK" and "Cancel". A dashed oval highlights the "SysLog Server" section.

**Figure 1-4: Logging with SysLog**

The SysLog Server resides in the MultiVOIP unit itself. To implement this functionality, you will need a SysLog client program (sometimes referred to as a "daemon"). SysLog client programs, both paid and freeware, can be obtained from Kiwi Enterprises, among other firms. See [www.kiwisyslog.com](http://www.kiwisyslog.com). SysLog client programs essentially give you a means of structuring console messages for convenience and ease of use.

MultiTech Systems does not endorse any particular SysLog client program. SysLog client programs by any qualified provider should suffice for use with MultiVOIP units. Kiwi's brief description of their SysLog program indicates the typical scope of such programs. "Kiwi Syslog Daemon is a freeware Syslog Daemon for the Windows platform. It receives, logs, displays and forwards Syslog messages from hosts such as routers, switches, Unix hosts and any other syslog enabled device. There are many customizable options available."

### MultiVOIP Front Panel LEDs

**LED Types.** The MultiVOIPs have two types of LEDs on their front panels:

- (1) general operation LED indicators (for power, booting, and ethernet functions), and
- (2) channel operation LED indicators that describe the data traffic and performance in each VOIP data channel.

**Active LEDs.** On both the MVPFX2-4 and MVPFX2-8, there four WAN LEDs and eight channel-operation LEDs. However, on the MVPFX2-4, only the left four sets of channel-operation LEDs are functional. On the MVPFX2-8, all eight sets are functional.



Figure 1-5. MVPFX2-8 LEDs

Similarly, the MVPFX2-2 has four WAN/Ethernet general-operation indicator LEDs and two channel-operation LEDs, one for each channel.

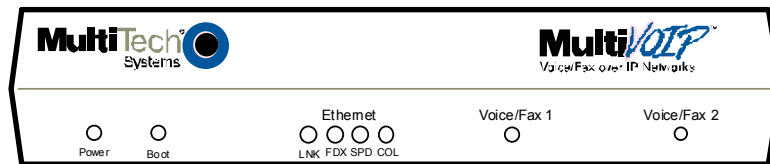


Figure 1-6. MVPFX2-2 LEDs

## LED Descriptions for MultiVOIP-FX2 Units

Front Panel LED Definitions	
LED NAME	DESCRIPTION
<b>General Operation LEDs</b> (one set on each MultiVOIP model)	
<b>Power</b>	Indicates presence of power.
<b>Boot</b>	After power up, the Boot LED will be on briefly while the MultiVOIP is booting. It lights whenever the MultiVOIP is booting or downloading a setup configuration data set.
<b>Ethernet</b>	<p><b>FDX.</b> LED indicates whether Ethernet connection is half-duplex or full-duplex (FDX) and, in half-duplex mode, indicates occurrence of data collisions. LED is on constantly for full-duplex mode; LED is off constantly for half-duplex mode. When operating in half-duplex mode, the LED will flash during data collisions.</p> <p><b>LNK.</b> Link/ Activity LED. This LED is lit if Ethernet connection has been made. It is off when the link is down (i.e., when no Ethernet connection exists). While link is up, this LED will flash off to indicate data activity.</p> <p><b>SPD.</b> Data speed indicator. When lit, data rate is 100 Mbps. When not lit, data rate is 10 Mbps.</p> <p><b>COL.</b> Collision indicator. Lit when data collision is detected on Ethernet network.</p>
<b>Channel-Operation LEDs</b>	
<b>Voice/Fax 1</b>	<i>On MVPFX2-2 only.</i> This indicator is lit when there is call activity on Channel 1. If the voip channel is operating as FXS, the LED is ON when the device attached to the channel is off hook. If the voip channel is operating as FXO, the LED is ON when the voip has seized the line.
<b>Voice/Fax 2</b>	<i>On MVPFX2-2 only.</i> This indicator is lit when there is call activity on Channel 2. If the voip channel is operating as FXS, the LED is ON when the device attached to the channel is off hook. If the voip channel is operating as FXO, the LED is ON when the voip has seized the line.
<b>1, 2, 3, ... 8</b>	<i>On MVPFX2-4 and MVPFX2-8.</i> There is one LED for each voip channel. The indicator for any channel is lit when there is call activity on that voip channel. If the voip channel is operating as FXS, the LED is ON when the device attached to the channel is off hook. If the voip channel is operating as FXO, the LED is ON when the voip has seized the line.

## Computer Requirements

The command computer used in conjunction with the MultiVOIP must meet these requirements:

- (a) any reasonably modern PC,
- (b) must have an up-to-date version of Java installed (v. 1.5 or higher),
- (c) must have an up-to-date web browser installed (at this writing, up-to-date browsers would include Internet Explorer 6.0(+), Netscape 6.0(+), or Mozilla FireFox 1.0(+).),
- (d) must have IP access to the MultiVOIP, and
- (e) optionally, have an available serial COM port for a console connection to the MultiVOIP.

This PC will generally be in contact with the MVPFX2 unit via the web. The Console connection, which requires a cable directly between the PC and the MultiVOIP is, essentially, a backup method of connecting to the voip. This direct connection can be used to reset the MultiVOIP's IP address and to upgrade firmware. The direct connection is not involved in the general operation of the MultiVOIP unit.

## Specifications

Parameter /Model	MVPFX2-4	MVPFX2-8	MVPFX2-2
<b>Operating Voltage/ Current</b>	100-240 VAC, 1.2 - 0.6 A	100-240 VAC 1.2 - 0.6 A	9 VDC, 1 A, External power supply
<b>Mains Frequencies</b>	50/60 Hz	50/60 Hz	50/60 Hz
<b>Power Consumption</b>	16 watts	26 watts	10 watts
<b>Mechanical Dimensions</b>	1.75" H x 17.4" W x 8.5" D  4.5cm H x 44.2 cm W x 21.6 cm D	1.75" H x 17.4" W x 8.5" D  4.5cm H x 44.2 cm W x 21.6 cm D	6.2" W x 9" D x 1.4" H  15.8cm W x 22.9cm D x 3.6cm H
<b>Weight</b>	7.1 lbs. (3.2 kg) includes power supply	7.24 lbs. (3.5 kg) includes power supply	1.8lbs (.82kg) 2.6lbs (1.17kg) without supply
<b>Operating Temperature</b>	0° to +60°C (32° to +120°F); humidity range 20-90% (non-condensing)	0° to +60°C (32° to +120°F); humidity range 20-90% (non-condensing)	0° to +60°C (32° to +120°F); humidity range 20-90% (non-condensing)
<b>Storage Temperature</b>	-10°C to +85°C	-10°C to +85°C	-10°C to +85°C

## Installation at a Glance

The basic steps of installing your MultiVOIP network involve unpacking the units, connecting the cables, and configuring the units using the MultiVOIP web-based graphic user interface (GUI), and confirming connectivity with another voip site. This process results in a fully functional Voice-Over-IP network.

## Related Documentation

The MultiVOIP User Guide (the document you are now reading) comes in electronic form and is included on your system CD. It presents in-depth information on the features and functionality of Multi-Tech's MultiVOIP Product Family. The MultiVOIP Cabling Guide, a printed document, is shipped with each MVPFX2 unit.

The CD media is produced using Adobe Acrobat™ for viewing and printing the user guide. To view or print your copy of a user guide, load Acrobat Reader™ on your system. The Acrobat Reader is included on the MultiVOIP CD and is also a free download from Adobe's Web Site:

[www.adobe.com/prodindex/acrobat/readstep.html](http://www.adobe.com/prodindex/acrobat/readstep.html)

This MultiVOIP User Guide is also available on Multi-Tech's Web site at:

<http://www.multitech.com>

Viewing and printing a user guide from the Web also requires that you have the Acrobat Reader loaded on your system. To select the MultiVOIP User Guide from the Multi-Tech Systems home page, click **Documents** and then click **MultiVOIP Family** in the product list drop-down window. All documents for this MultiVOIP Product Family will be displayed. You can then choose *User Guide (MultiVOIP Product Family)* to view or download the **.pdf** file.

Entries (organized by model number) in the "knowledge base" and 'troubleshooting resolutions' sections of the MultiTech web site (found under "Support") constitute another source of help for problems encountered in the field.

---

## **Chapter 2: Quick Start Guide**

This chapter contains streamlined instructions to get the MultiVOIP up and running quickly. These start-up instructions include assistance on setting up the MultiVOIP's Inbound and Outbound Phonebooks. These sections of the Quick Start Guide may be particularly useful for phonebook configuration:

- Phonebook Starter Configuration
- Phonebook Tips
- Phonebook Example (One Common Situation)

The Quick Start Guide also contains a "Phonebook Worksheet" section. You may want to print out several worksheet copies. Paper copies can be very helpful in comparing phonebooks at multiple sites at a glance. This will assist you in making the phonebooks clear and consistent and will reduce 'surfing' between screens on the configuration program.

A printed Cabling Guide is shipped with the MultiVOIP and an electronic copy is included on the Product CD.


## MultiVOIP Startup Tasks

Task	Summary
● <b>Collecting Phone/IP Details ( vital! )</b>	The MultiVOIP must be configured to interface with your particular phone system and IP network. To do so, certain details must be known about those phone and IP systems.
● <b>Command/Control Computer Setup: Specs &amp; Settings</b>	Some modest minimum specifications must be met. A COM port must be set up.
● <b>Placement</b>	Decide where you'll mount the voip.
● <b>Hookup</b>	Connect power, phone, and data cables per the <i>Quick Hookup</i> diagrams in this chapter.
● <b>Software Installation</b>	Check that an up-to-date version of Java (version 1.5) is on your computer. If not, install it from the MultiVOIP CD or the Java website.
● <b>Phone/IP Starter Configuration</b>	You will enter phone numbers and IP addresses. You'll use default parameter values where possible to get the system running quickly. Use "Config Info CheckList" (page 17).
● <b>Phonebook Starter Configuration</b>	The phonebook is where you specify how calls will be routed. To get the system running quickly, you'll make phonebooks for just two voip sites.
● <b>Connectivity Test</b>	You'll find out if your voip system can carry phone calls between two sites. That means you're up and running!
● <b>Troubleshooting</b>	Detect and remedy any problems that might have prevented connectivity.

## Phone/IP Details \*Absolutely Needed\* Before Starting the Installation


The MultiVOIP will interface with both the IP network and the phone system. You must gather information about the IP network and about the phone system so that the MultiVOIP can be configured to operate with them properly. A summary of this configuration information appears on page 17 (“Config Info CheckList”).

### Gather IP Information

➔	<i>Ask your computer network administrator.</i>	<i>Info needed to operate:</i> <b>all MultiVOIP models.</b>
	 IP Network Parameters: Record for each VOIP Site in System	
	• IP Address	
	• IP Mask	
	• Gateway	
	• Domain Name Server (DNS) Info (optional)	

## Phone/IP Details \*Absolutely Needed\*

### Gather Telephone Information (Analog)

➔	<b>Analog Phone Parameters</b>  <i>Ask phone company or telecom manager.</i>	<i>Needed for:</i> <b>MVPFX2-8</b> <b>MVPFX2-4</b> <b>MVPFX2-2</b>
	 Analog Telephony Interface Parameters: Record for this VOIP Site	
	• Which interface type is used? FXS _____ FXO _____	
	• If FXS, determine whether the line will be used for a phone, fax, or KTS (key telephone system)	
	• If FXO, determine if line will be an analog PBX extension or an analog line from a telco central office	



### Config Info CheckList

Type of Config Info Gathered	MultiVOIP Configuration screen on which to enter Config Info	Info Obtained ✓	Info Entered
IP info for voip unit <ul style="list-style-type: none"> <li>• IP address</li> <li>• Gateway</li> <li>• DNS IP (if used)</li> </ul>	IP Parameters		
Interface Type (Choices: FXS or FXO*)	Interface Parameters *In FXO/FXS systems, channels used for phone, fax, or key system are FXS; channels used for analog PBX extensions or analog telco lines are FXO.		
Country Code	Regional Parameters		
<b>Network Locations</b> of SIP Proxy units, if used (IP Address or Domain Name)	SIP Call Signaling		
<b>Reminder:</b> <i>Be sure to Save &amp; Apply after entering configuration values.</i>			

### Identify Remote VOIP Site to Call

When you're done installing the MultiVOIP, you'll want to confirm that it is configured and operating properly. To do so, it's good to have another voip that you can call for testing purposes. You'll want to confirm end-to-end connectivity. You'll need IP and telephone information about that remote site.

If this is the very first voip in the system, you'll want to coordinate the installation of this MultiVOIP with an installation of another unit at a remote site.

## Command/Control Computer Setup (Specs & Settings)

The computer used for command and control of the MultiVOIP

- (a) any reasonably modern PC,
- (b) must have an up-to-date version of Java installed (v. 1.5 or higher),
- (c) must have an up-to-date web browser installed (at this writing, up-to-date browsers would include Internet Explorer 6.0(+), Netscape 6.0(+), or Mozilla FireFox 1.0(+).),
- (d) must have IP access to the MultiVOIP, and
- (e) optionally, have an available serial COM port for a console connection to the MultiVOIP.

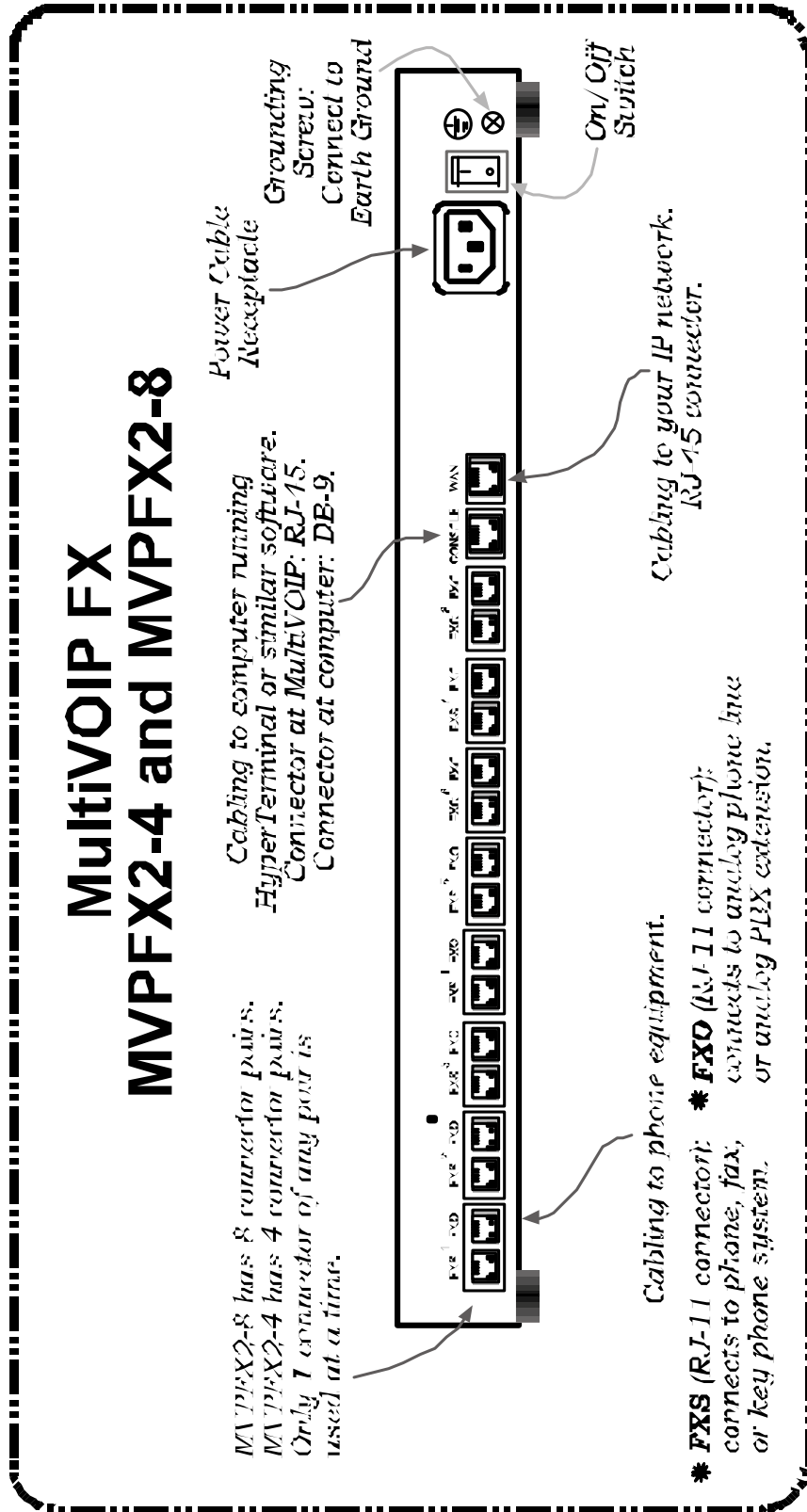
The configuration tasks and control tasks the PC will have to do with the MultiVOIP are not especially demanding. Still, we recommend using a reasonably new computer. The computer that you use to configure your MultiVOIP need not be dedicated to the MultiVOIP after installation is complete.

**COM port on controller PC.** If you choose to use the MultiVOIP's Console connection, you will need an available COM port on the controller PC. You'll need to know which COM port is available for use with the MultiVOIP (COM1, COM2, etc.).

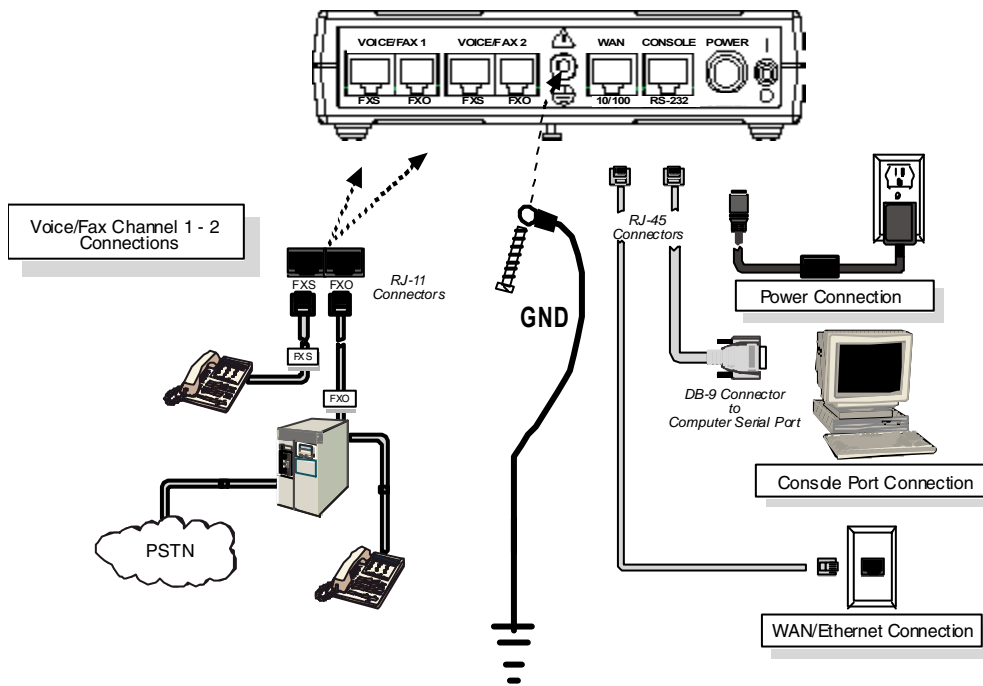
## Placement

Mount your MultiVOIP in a safe and convenient location where cables for your network and phone system are accessible. Rack-mounting instructions are in *Chapter 3: Mechanical Installation & Cabling*.

### Quick Hookup for MVPFX2-4 & MVPFX2-8



### Quick Hookup for MVPFX2-2



## Ensure that Java & Browser Versions will Support Web-Based GUI

For more details, see *Chapter 4: Software Installation* in User Guide.

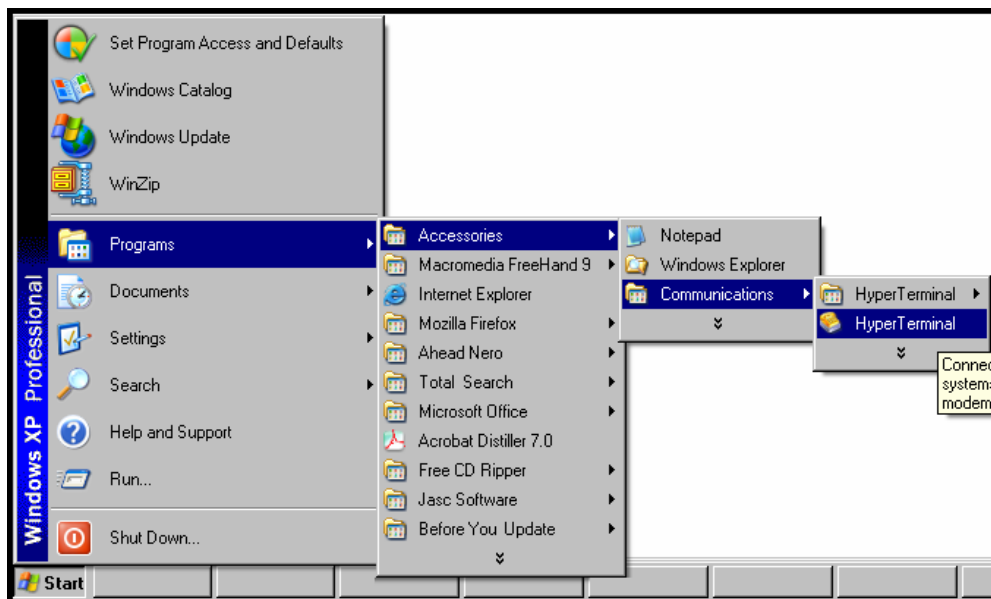
1. MultiVOIP must be properly cabled. Power must be turned on.
2. Is Java Runtime program at level 1.5 or greater? If not, load up-to-date Java version from MultiVOIP CD or from Java web site.
3. Is web browser of a sufficiently recent version to support MultiVOIP web GUI? (The browser must be Internet Explorer 6.0(+), Netscape 6.0(+), or FireFox 1.0(+).) If not, download a browser version that is new enough to support the web GUI.
4. Browse to IP address of MultiVOIP unit (default is 192.168.2.1).
5. If username and password have been established, enter them when prompted by voip.
6. Use web browser GUI to continue with configuration and operation of voip.

## Changing the IP Address through the Console Connection

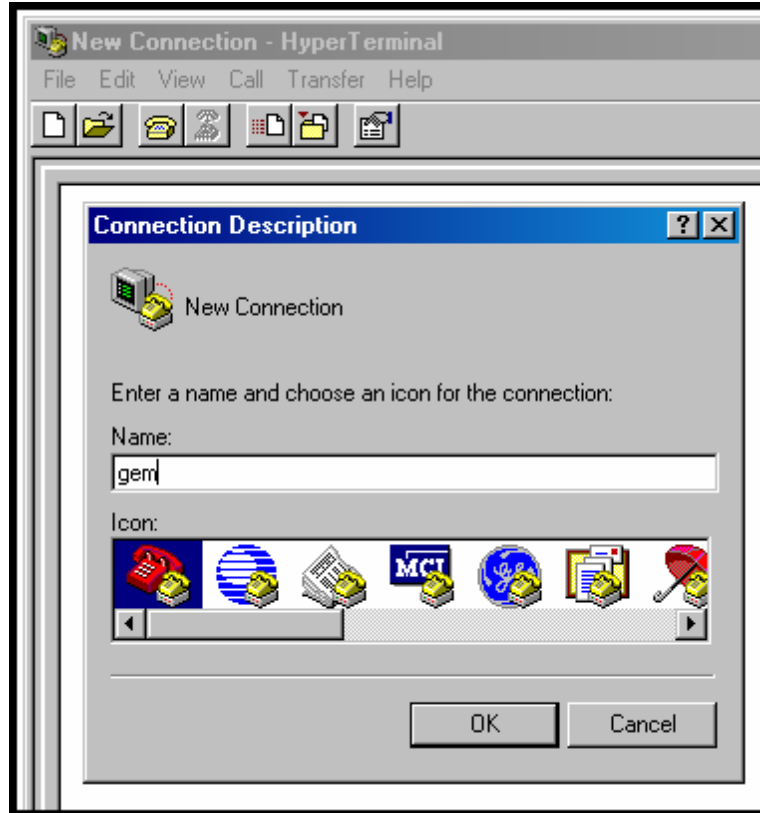
At its initial bootup, the default IP address of the MultiVOIP is 192.168.2.1. If you are not able to access the web GUI through this IP address (192.168.2.1), then use the procedure below to set a valid IP for operation of the MultiVOIP on your network.

This procedure also works if the IP address is forgotten.

1. Connect a cable between the MultiVOIP’s “Console” connector and a serial cable on the computer.
2. Launch HyperTerminal or a similar communications program.



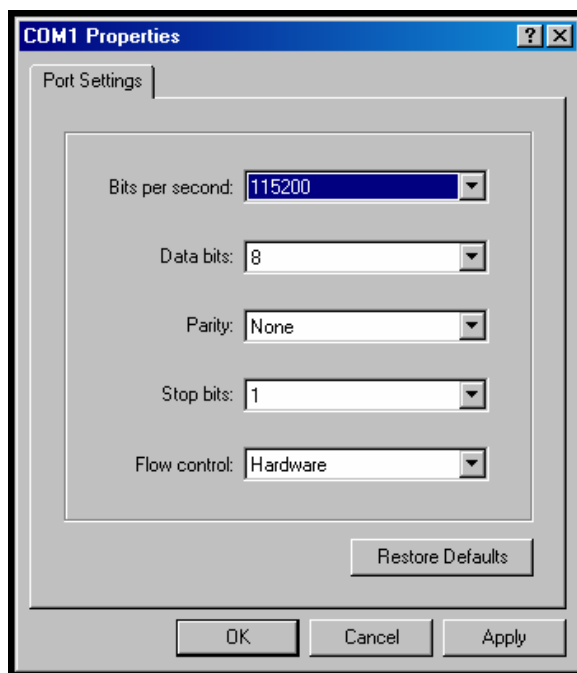
3. Establish a 'connection' in HyperTerminal.



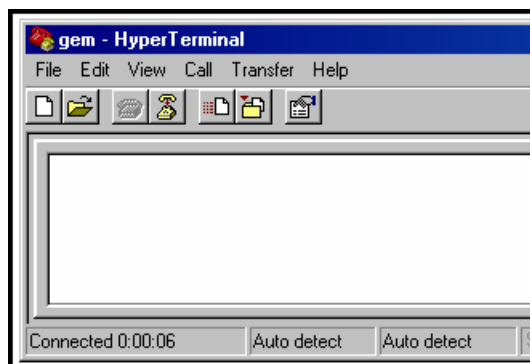
4. Check that HyperTerminal is addressing the correct COM port.



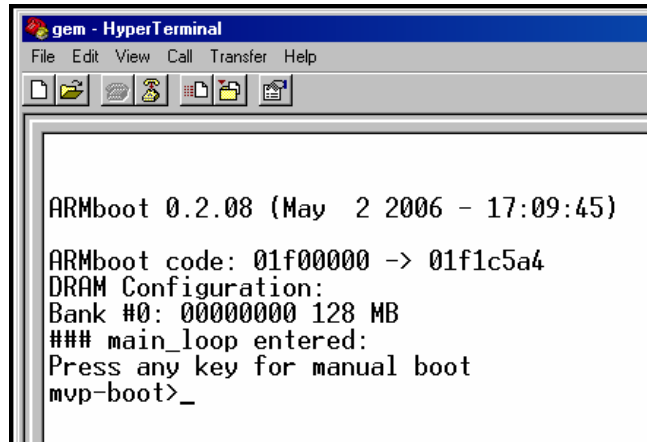
5. Check that HyperTerminal's data rate is set to 115200bps.



6. To begin, HyperTerminal must be connected and ready.



7. Reboot the MultiVOIP by turning off its power and turning it back on again.  
The ARMBoot prompt will appear on the HyperTerminal screen.



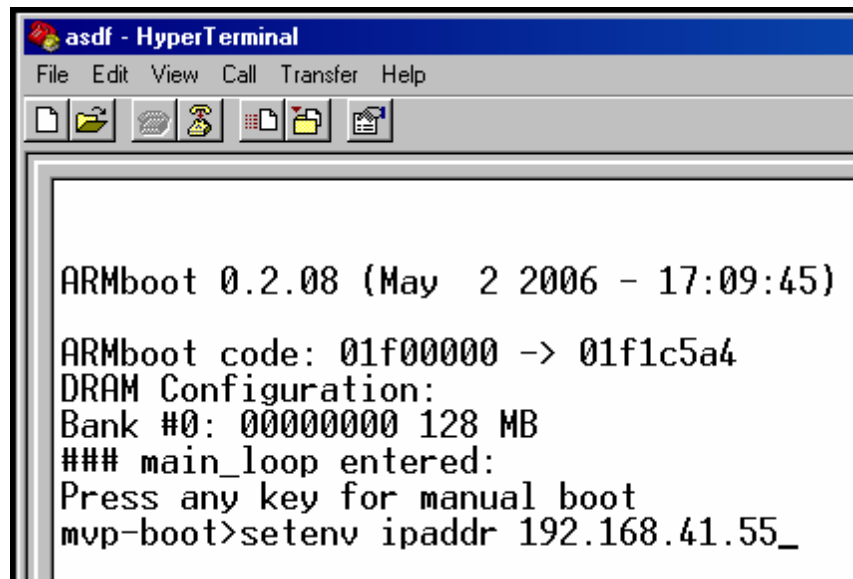
```
gem - HyperTerminal
File Edit View Call Transfer Help

ARMboot 0.2.08 (May  2 2006 - 17:09:45)

ARMboot code: 01f00000 -> 01f1c5a4
DRAM Configuration:
Bank #0: 00000000 128 MB
### main_loop entered:
Press any key for manual boot
mvp-boot>_
```

When this screen appears, you must quickly press any key to stop the regular boot-up process (the manual boot process).

8. Type `setenv ipaddr a.b.c.d` where a, b, c, & d are the octet values for the desired IP address of the voip.



```
asdf - HyperTerminal
File Edit View Call Transfer Help

ARMboot 0.2.08 (May  2 2006 - 17:09:45)

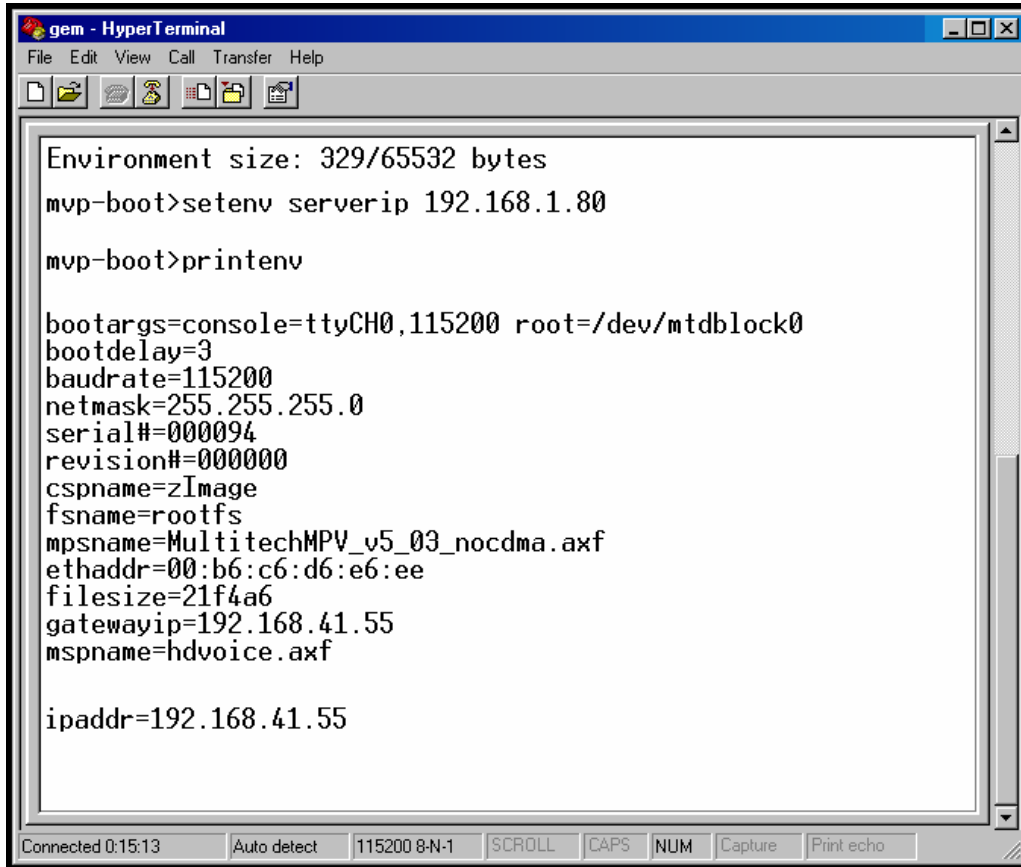
ARMboot code: 01f00000 -> 01f1c5a4
DRAM Configuration:
Bank #0: 00000000 128 MB
### main_loop entered:
Press any key for manual boot
mvp-boot>setenv ipaddr 192.168.41.55_
```

Press **Enter**.

**Note:** When using the `setenv` command, be careful in your spelling. If you mis-spell `ipaddr` as “ipadde” for example, the ARMBoot program will create a new and useless variable entitled `ipadde` and will not change the value of the `ipaddr` variable.



9. To confirm that the **ipaddr** (voip IP address) was indeed changed to the value you want, type **printenv** at the **mvp-boot>** prompt and then press **Enter**.



```

gem - HyperTerminal
File Edit View Call Transfer Help
Environment size: 329/65532 bytes
mvp-boot>setenv serverip 192.168.1.80
mvp-boot>printenv

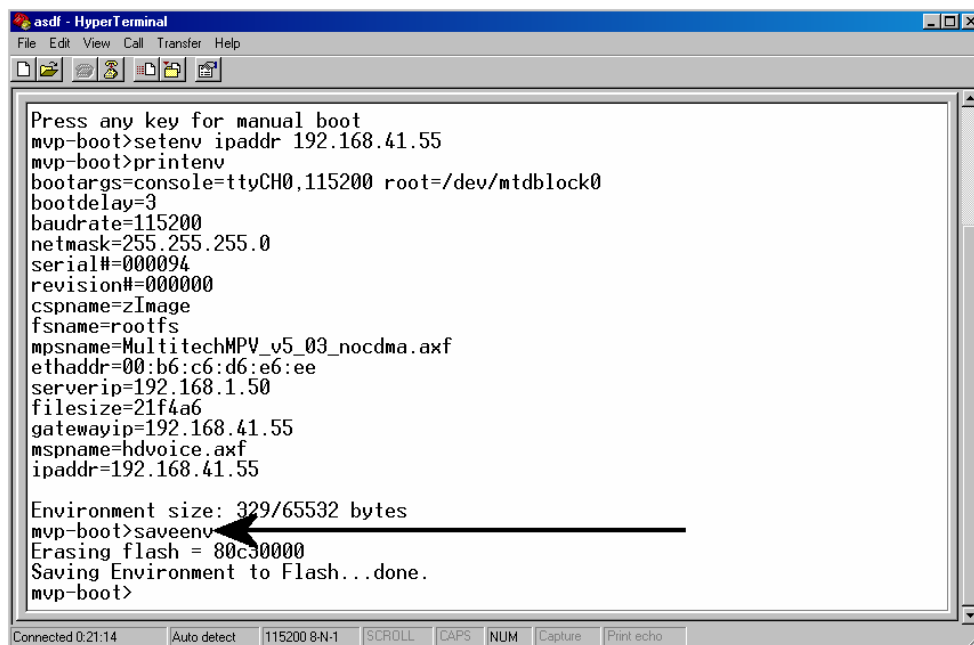
bootargs=console=ttyCH0,115200 root=/dev/mtdblock0
bootdelay=3
baudrate=115200
netmask=255.255.255.0
serial#=000094
revision#=000000
cspname=zImage
fsname=rootfs
mpsname=MultitechMPV_v5_03_nocdma.axf
ethaddr=00:b6:c6:d6:e6:ee
filesize=21f4a6
gatewayip=192.168.41.55
mstype=hdvoice.axf

ipaddr=192.168.41.55

Connected 0:15:13  Auto detect  115200 8-N-1  SCROLL  CAPS  NUM  Capture  Print echo

```

10. Type **saveenv** and press **Enter**.



```

asdf - HyperTerminal
File Edit View Call Transfer Help
Press any key for manual boot
mvp-boot>setenv ipaddr 192.168.41.55
mvp-boot>printenv
bootargs=console=ttyCH0,115200 root=/dev/mtdblock0
bootdelay=3
baudrate=115200
netmask=255.255.255.0
serial#=000094
revision#=000000
cspname=zImage
fsname=rootfs
mpsname=MultitechMPV_v5_03_nocdma.axf
ethaddr=00:b6:c6:d6:e6:ee
serverip=192.168.1.50
filesize=21f4a6
gatewayip=192.168.41.55
mstype=hdvoice.axf
ipaddr=192.168.41.55

Environment size: 329/65532 bytes
mvp-boot>saveenv
Erasing flash = 80c30000
Saving Environment to Flash...done.
mvp-boot>

```

11. Turn the voip off and then on again to reboot it.

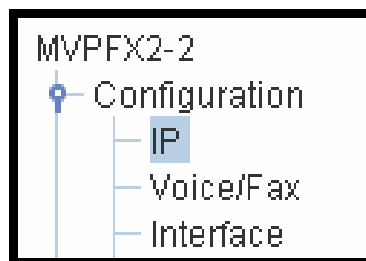
- Allow the voip to boot up again normally (this will take a few minutes) with the console connection still active. When the rebooting process is complete (when the boot light is no longer lit), type **1** and press **Enter**.

**NOTE:** This change of IP address is only temporary. You must complete this procedure to make the change of IP address permanent.

```

[000000007470] [INFO] STIF *****
1
bootargs=console=ttyCH0,115200 root=/dev/mtdblock0
bootdelay=3
baudrate=115200
netmask=255.255.255.0
serial#=000094
revision#=000000
cspname=zImage
fsname=rootfs
mbspname=MultitechMPV_v5_03_nocdma.axf
ethaddr=00:b6:c6:d6:e6:ee
serverip=192.168.1.50
filesize=21f4a6
gatewayip=192.168.41.55
mbspname=hdvoice.axf
ipaddr=192.168.41.55
ipaddr=192.168.41.55
[000000150349] [INFO] MVPAPP setIneterfaceIP() for interface eth0: 192.168.41.5
5
[000000150359] [INFO] MVPAPP Success Setting IP address
gatewayip=192.168.41.55
[000000150379] [INFO] MVPAPP Success: in Setting Gateway address 192.168.41.55
    
```

- Use a web browser to browse to the voip using the IP address that you have just assigned.
- In the web browser, click on **IP Parameters** in the sidebar list.



In each field, enter the values that fit your particular network.

Current Permission: Read/Write

### IP Parameters

#### Diffserv Parameters

Call Control PHB

VoIP Media PHB

Type II ▼

#### IP Parameters

Gateway Name

Enable DHCP

IP Address

IP Mask

Gateway

#### DNS

Enable DNS

DNS Server IP Address

#### FTP Server

Enable

OK

Cancel

Click **OK**.

15. In the sidebar menu, click **Save & Apply**. Allow the voip to Reboot.

## Phone/IP Starter Configuration

Full details here:

MVPFX2-2 MVPFX2-4 MVPFX2-8	<i>Technical Configuration</i> chapter in <i>User Guide</i>
----------------------------------	--

1. Open a browser and go to the IP Address of the MVPFX2 unit. In the sidebar menu click **Configuration**.
2. Go to **Configuration | IP**. Enter or alter any IP Parameters, as needed. Click **OK**.
3. Go to **Configuration | Voice/Fax**. Select **Coder | "Automatic."** At the right-hand side of the dialog box, click **OK**. If you know any specific parameter values that will apply to your system, enter them. Click **Copy Channel**. Select **Copy to All**. Click **Copy**. At main Voice/Fax Parameters screen, click **OK** to exit from the dialog box.
4. Enter telephone system information. Go to **Configuration | Interface**. Enter parameters obtained from phone company or PBX administrator. Click **OK**.
5. Go to **Configuration | Regional Parameters**. Select the **Country/Region** that fits your situation. Click **OK** and confirm. Click **OK** to exit from the dialog box.
6. Go to **Configuration | Logs/Traces**.  
Select "Enable Console Messages." Click **OK**.  
  
To do logging with a SysLog client program, click on "SysLog Server - Enable" in the **Logs/Traces** screen. To implement this function, you must install a SysLog client program. For more info, see the "SysLog Server Functions" section of the *Operation & Maintenance* chapter of the **User Guide**.
7. Go to **Save &Apply**. Click **OK**. This will save the parameter values that you have just entered.

The MultiVOIP's "BOOT" LED will light up while the configuration file is being saved and loaded into the MultiVOIP. Don't do anything to the MultiVOIP until the "BOOT" LED is off (a loss of power at this point could cause the MultiVOIP unit to lose the configuration settings you have made).

END OF PROCEDURE.

## Phonebook Starter Configuration (*with remote voip*)

If the topic of voip phone books is new to you, it may be helpful to read the PhoneBook Tips section (page 34) before starting this procedure.

To do this part of the quick setup, you need to know of another voip that you can call to conduct a test. Ideally, a test of two voips at the same physical location connected back-to-back should be done first. A secondary test should be done between two voips at different locations, typically with one voip located somewhere outside of your building. You must know the phone number and IP address for that site. We are assuming here that the MultiVOIP will operate in conjunction with a PBX.

You must configure both the Outbound Phonebook and the Inbound Phonebook. A starter configuration only means that two voip locations will be set up to begin the system and establish voip communication.

### Outbound Phonebook

1. Open the browser and go to the IP address of the MultiVOIP unit. In the sidebar menu, select **Phone Book**
2. Go to **Outbound Phonebook | Add Entry**.
3. On a sheet of paper, write down the calling code of the remote voip (area code, country code, city code, etc.) that you'll be calling.

Follow the example that best fits your situation.

<p style="text-align: center;"><b>North America, Long-Distance Example</b></p> <p>Technician in Seattle (area 206) must set up one voip there, another in Chicago (area 312, downtown).</p> <p><b>Answer:</b> Write down <b>312</b>.</p>	<p style="text-align: center;"><b>Euro, National Call Example</b></p> <p>Technician in central London (area 0207) to set up voip there, another in Birmingham (area 0121).</p> <p><b>Answer:</b> write down <b>0121</b>.</p>
<p style="text-align: center;"><b>Euro, International Call Example</b></p> <p>Technician in Rotterdam (country 31; city 010) to set up one voip there, another in Bordeaux (country 33; area 05).</p> <p><b>Answer:</b> write down <b>3305</b>.</p>	

4. Suppose you want to call a phone number outside of your building using a phone station that is an extension from your PBX system (if present). What digits must you dial? Often a “9” or “8” must be dialed to “get an outside line” through the PBX (i.e., to connect to the PSTN). Generally, “1” or “11” or “0” must be dialed as a prefix for calls outside of the calling code area (long-distance calls, national calls, or international calls).

On a sheet of paper, write down the digits you must dial before you can dial a remote area code.

<p style="text-align: center;"><b>North America, Long-Distance Example</b></p> <p>Seattle-Chicago system.</p> <p>Seattle voip works with PBX that uses “8” for all voip calls. “1” must immediately precede area code of dialed number.</p> <p><b>Answer:</b> write down <b>81</b>.</p>	<p style="text-align: center;"><b>Euro, National Call Example</b></p> <p>London/Birming. system.</p> <p>London voip works with PBX that uses “9” for all out-of-building calls whether by voip or by PSTN. “0” must immediately precede area code of dialed number.</p> <p><b>Answer:</b> write down <b>90</b>.</p>
<p style="text-align: center;"><b>Euro, International Call Example</b></p> <p>Rotterdam/Bordeaux system.</p> <p>Rotterdam voip works with PBX where “9” is used for all out-of-building calls. “0” must precede all international calls.</p> <p><b>Answer:</b> write down <b>90</b>.</p>	

5. In the “Destination Pattern” field of the **Outbound Phone Book Add Entry** screen, enter the digits from step 4 followed by the digits from step 3.

<p style="text-align: center;"><b>North America, Long-Distance Example</b></p> <p>Seattle-Chicago system.</p> <p><b>Answer:</b> enter <b>81312</b> as Destination Pattern in Outbound Phone-book of Seattle voip.</p>	<p style="text-align: center;"><b>Euro, National Call Example</b></p> <p>London/Birming. system.</p> <p>Leading zero of Birmingham area code is dropped when combined with national-dialing access code. (Such practices vary by country.)</p> <p><b>Answer:</b> enter <b>90121</b> as Destination Pattern in Outbound Phonebook of London voip. <i>Not 900121.</i></p>
<p style="text-align: center;"><b>Euro, International Call Example</b></p> <p>Rotterdam/Bordeaux system.</p> <p><b>Answer:</b> enter <b>903305</b> as Destination Pattern in Outbound Phonebook of Rotterdam voip.</p>	

6. In the "Remove Prefix" field, enter the initial PBX access digit ("8" or "9").

<p><b>North America, Long-Distance Example</b> Seattle-Chicago system. <b>Answer:</b> enter <b>8</b> in "Remove Prefix" field of Seattle Outbound Phonebook.</p>	<p><b>Euro, National Call Example</b> London/Birming. system. <b>Answer:</b> enter <b>9</b> in "Remove Prefix" field of London Outbound Phonebook.</p>
<p><b>Euro, International Call Example</b> Rotterdam/Bordeaux system. <b>Answer:</b> enter <b>9</b> in "Remove Prefix" field of Outbound Phonebook for Rotterdam voip.</p>	

Some PBXs will not 'hand off' the "8" or "9" to the voip. But for those PBX units that do, it's important to enter the "8" or "9" in the "Remove Prefix" field in the Outbound Phonebook. This precludes the problem of having to make two inbound phonebook entries at remote voips, one to account for situations where "8" is used as the PBX access digit, and another for when "9" is used.

- 7. If you intend to use a SIP Proxy, enter the relevant information in the Call Signaling screen.
- 8. Enter the IP address of the MultiVOIP that you want to call.
- 9. Click **OK.** to exit from the **Outbound Phonebook Add Entry** screen.

### Inbound Phonebook

1. Open the browser and go to the IP address of the MultiVOIP unit.
2. Go to **Phone Book | Inbound Phonebook | Add Entry**.
3. In the "Remove Prefix" field, enter your local calling code (area code, country code, city code, etc.) preceded by any other "access digits" that are required to reach your local site from the remote voip location (think of it as though the call were being made through the PSTN - even though it will not be).

<p><b>North America, Long-Distance Example</b></p> <p>Seattle-Chicago system.</p> <p>Seattle is area 206. Chicago employees must dial 81 before dialing any Seattle number on the voip system.</p> <p>Answer: <b>1206</b> is prefix to be removed by local (Seattle) voip.</p>	<p><b>Euro, National Call Example</b></p> <p>London/Birming. system.</p> <p>Inner London is 0207 area. Birmingham employees must dial 9 before dialing any London number on the voip system.</p> <p>Answer: <b>0207</b> is prefix to be removed by local (London) voip.</p>
<p><b>Euro, International Call Example</b></p> <p>Rotterdam/Bordeaux system.</p> <p>Rotterdam is country code 31, city code 010. Bordeaux employees must dial 903110 before dialing any Rotterdam number on the voip system.</p> <p>Answer: <b>03110</b> is prefix to be removed by local (Rotterdam) voip.</p>	



4. In the "Add Prefix" field, enter any digits that must be dialed from your local voip to gain access to the PSTN.

<p><b>North America, Long-Distance Example</b></p> <p>Seattle-Chicago system.</p> <p>On Seattle PBX, "9" is used to get an outside line.</p> <p>Answer: <b>9</b> is prefix to be added by local (Seattle) voip.</p>	<p><b>Euro, National Call Example</b></p> <p>London/Birming. system.</p> <p>On London PBX, "9" is used to get an outside line.</p> <p>Answer: <b>9</b> is prefix to be added by local (London) voip.</p>
<p><b>Euro, International Call Example</b></p> <p>Rotterdam/Bordeaux system.</p> <p>On Rotterdam PBX, "9" is used to get an outside line.</p> <p>Answer: <b>9</b> is prefix to be added by local (Rotterdam) voip.</p>	

5. In the "Channel Number" field, enter "Hunting." A "hunting" value means the voip unit will assign the call to the first available channel. If desired, specific channels can be assigned to specific incoming calls (i.e., to any set of calls received with a particular incoming dialing pattern).

6. In the "Description" field, it is useful to describe the ultimate destination of the calls. For example, in a New York City voip system, "incoming calls to Manhattan office," might describe a phonebook entry, as might the descriptor "incoming calls to NYC local calling area." The description should make the routing of calls easy to understand. (40 characters max.)

<p><b>North America, Long-Distance Example</b></p> <p>Seattle-Chicago system.</p> <p>Possible Description: Free Seattle access, all employees</p>	<p><b>Euro, National Call Example</b></p> <p>London/Birming. system.</p> <p>Possible Description: Local-rate London access, all empl.</p>
<p><b>Euro, International Call Example</b></p> <p>Rotterdam/Bordeaux system.</p> <p>Possible Description: Local-rate Rotterdam access, all empl.</p>	

7. Repeat steps 2-6 for each inbound phonebook entry. When all entries are complete, go to step 8.
8. Click **OK** to exit the inbound phonebook screen.
9. Click on **Save & Apply**. Click **OK**. Then click **Reboot Voip**.
- Your starter inbound phonebook configuration is complete.

## Phonebook Tips

Preparing the phonebook for your voip system is a complex task that, at first, seems quite daunting. These tips may make the task easier.

**1. Use Dialing Patterns, Not Complete Phone Numbers.** You will not generally enter complete phone numbers in the voip phonebook. Instead, you'll enter "destination patterns" that involve area codes and other digits. If the destination pattern is a whole area code, you'll be assigning all calls to that area code to go to a particular voip which has a unique IP address. If your destination pattern includes an area code plus a particular local phone exchange number, then the scope of calls sent through your voip system will be narrowed (only calls within that local exchange will be handled by the designated voip, not all calls in that whole area code). In general, when there are fewer digits in your destination pattern, you are asking the voip to handle calls to more destinations.

**2. The Four Types of Phonebook Digits Used. Important!**

"Destination patterns" to be entered in your phonebook will generally consist of:

- (a) calling area codes,
- (b) access codes,
- (c) local exchange numbers, and
- (d) specialized codes.

Although voip phonebook entries may look confusing at first, it's useful to remember that all the digits in any phonebook entry must be of one of these four types.

(a) **calling area codes.** There are different names for these around the world: "area codes," "city codes," "country codes," etc. These codes, are used when making non-local calls. They always precede the phone number that would be dialed when making a local call.

(b) **access codes.** There are digits (*PSTN access codes*) that must be dialed to gain access to an operator, to access the publicly switched 'long-distance' calling system (North America), to access the publicly switched 'national' calling system (Europe and elsewhere), or to access the publicly switched 'international' calling system (worldwide).

There are digits (*PBX access codes*) that must be dialed by phones connected to PBX systems or key systems. Often a "9" must be dialed on a PBX phone to gain access to the PSTN ('to get an outside line'). Sometimes "8" must be dialed on a PBX phone to divert calls onto a leased line or to a voip system. However, sometimes PBX systems are 'smart' enough to route calls to a voip system without a special access code (so that "9" might still be used for all calls outside of the building).

There are also digits (*special access codes*) that must be dialed to gain access to a particular discount long-distance carrier or to some other closed or proprietary telephone system.

(c) **local exchange numbers.** Within any calling area there will be many local exchange numbers. A single exchange may be used for an entire small town. In cities, an exchange may be used for a particular neighborhood (although exchanges in cities do not always cover easily discernible areas). Organizations like businesses, governments, schools, and universities are also commonly assigned exchange numbers for their exclusive use. In some cases, these organizational-assigned exchanges can become non-localized because the exchange is assigned to one facility and linked, by the organization's private network, to other sometimes distant locations.

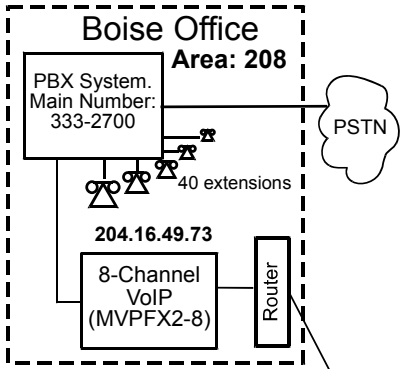
(d) **specialized codes.** Some proprietary voip units assign, to sites and phone stations, numbers that are not compatible with PSTN numbering. This can also occur in PBX or key systems. These specialized numbers must be handled on a case-by-case basis.



Phonebook Example

One Common Situation

**Voip Example.** This company has offices in three different cities. The PBX units all operate alike. Notably, they all give access to outside lines using “9.” They all are ‘smart’ enough to identify voip calls without using a special access digit (“8” is used in some systems). Finally, the system operates so that employees in any office can dial employees in any other office using only three digits. Here are the phonebooks needed for that system.

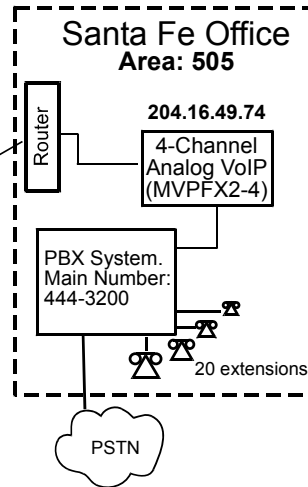


**Inbound Phonebook**

Each **Inbound Phonebook** contains two entries. The first entry (4 digits) specifies how incoming calls from the other voip sites will be handled if they go out onto the local PSTN. Essentially, all those calls come to the receiving voip with a pattern beginning with 1+area code. The local voip removes those four digits because they aren't needed when dialing locally. The local voip attaches a “9” at the beginning of the number to get an outside line. The PBX then completes the call to the PSTN.

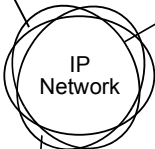
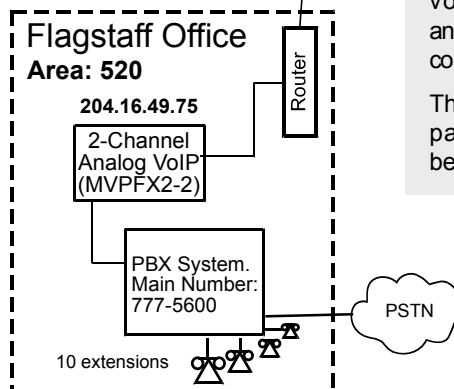
The second **Inbound Phonebook** entry (1 digit) is for receiving calls from company employees in the other two cities. The out-of-town employee simply dials 3 digits. The first of the three digits is uniquely used at each site and so acts as a destination pattern (Boise extensions are 7xx, Santa Fe extensions 2xx, Flagstaff extensions 6xx).

The local voip sees the pattern in its inbound phone book and notes the first digit (here either 2, 5, or 6). To make the match, this first digit, 2, 5, or 6 is put in the “Remove Prefix” field. This first digit must then be added back once again so that the voip will send all three digits to the PBX. The PBX can then dial the specific extension identified by the three-digit number.

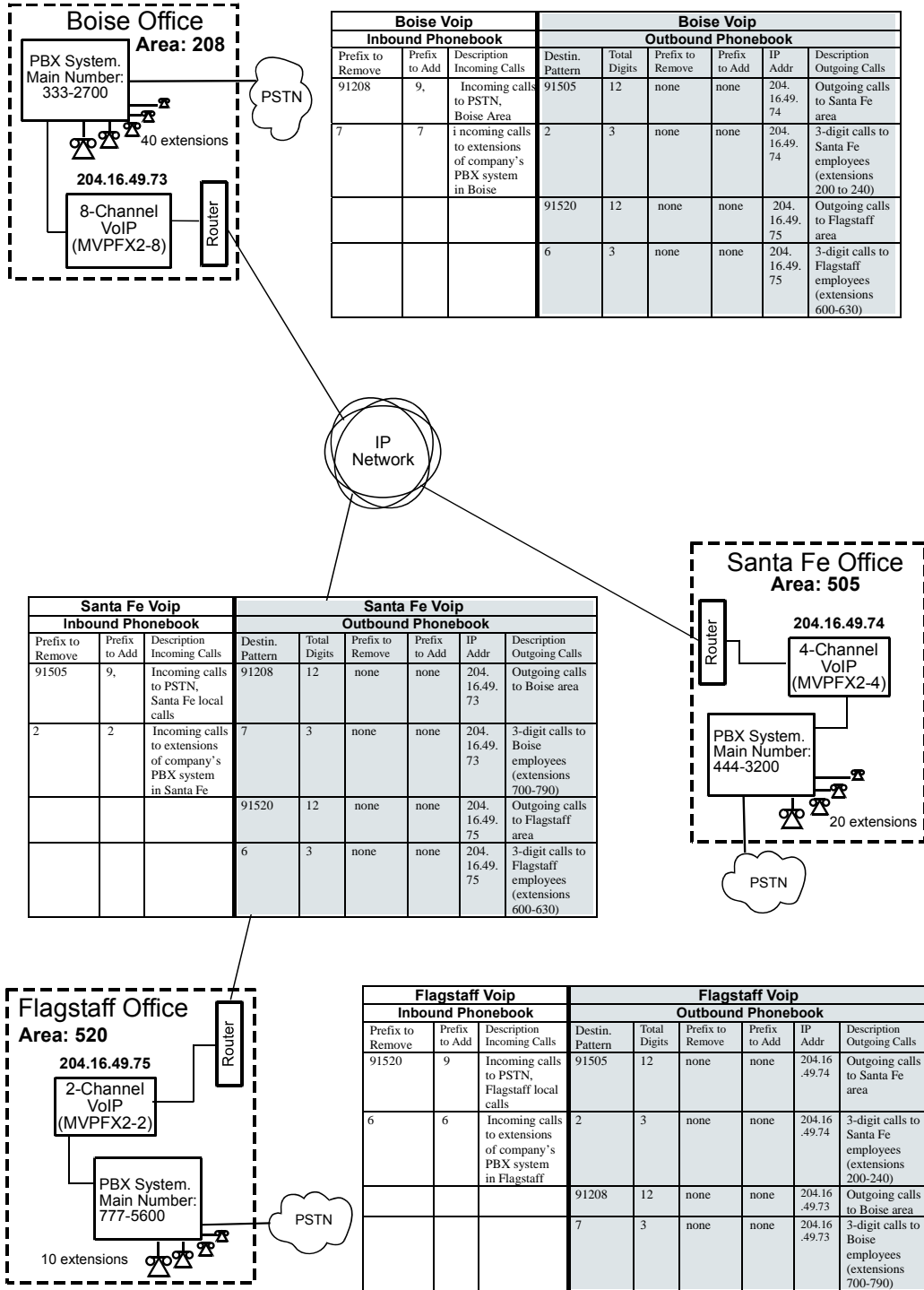


Each **Outbound Phonebook** contains two pairs of entries, two entries for each remote site. Whenever an out-of-town employee dials a 12-digit number beginning with the listed 5-digit destination pattern (9+ 1+area code) of another company location, the PBX hands the call to the voip system. The local voip strips off the “9” and directs the call to the IP address of the remote voip. The remote voip receives the call and hands it to its PBX. The PBX then completes the call to the PSTN.

The one-digit **Outbound** destination patterns pertain to 3-digit calling between company employees.



Voip Sites with Phonebooks



### Sample Phonebooks Enlarged

Boise Voip			Boise Voip					
Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
91208	9,	Incoming calls to PSTN, Boise Area	91505	12	none	none	204.16.49.74	Outgoing calls to Santa Fe area
7	7	incoming calls to extensions of company's PBX system in Boise	2	3	none	none	204.16.49.74	3-digit calls to Santa Fe employees (extensions 200 to 240)
			91520	12	none	none	204.16.49.75	Outgoing calls to Flagstaff area
			6	3	none	none	204.16.49.75	3-digit calls to Flagstaff employees (extensions 600-630)

Santa Fe Voip			Santa Fe Voip					
Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
91505	9,	Incoming calls to PSTN, Santa Fe local calls	91208	12	none	none	204.16.49.73	Outgoing calls to Boise area
2	2	Incoming calls to extensions of company's PBX system in Santa Fe	7	3	none	none	204.16.49.73	3-digit calls to Boise employees (extensions 700-790)
			91520	12	none	none	204.16.49.75	Outgoing calls to Flagstaff area
			6	3	none	none	204.16.49.75	3-digit calls to Flagstaff employees (extensions 600-630)

Flagstaff Voip			Flagstaff Voip					
Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
91520	9	Incoming calls to PSTN, Flagstaff local calls	91505	12	none	none	204.16.49.74	Outgoing calls to Santa Fe area
6	6	Incoming calls to extensions of company's PBX system in Flagstaff	2	3	none	none	204.16.49.74	3-digit calls to Santa Fe employees (extensions 200-240)
			91208	12	none	none	204.16.49.73	Outgoing calls to Boise area
			7	3	none	none	204.16.49.73	3-digit calls to Boise employees (extensions 700-790)

# Phonebook Worksheet

Voip Location/ID: \_\_\_\_\_

Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls

Other Details:

Voip Location/ID: \_\_\_\_\_

Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls

Other Details:

Voip Location/ID: \_\_\_\_\_

Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls

Other Details:





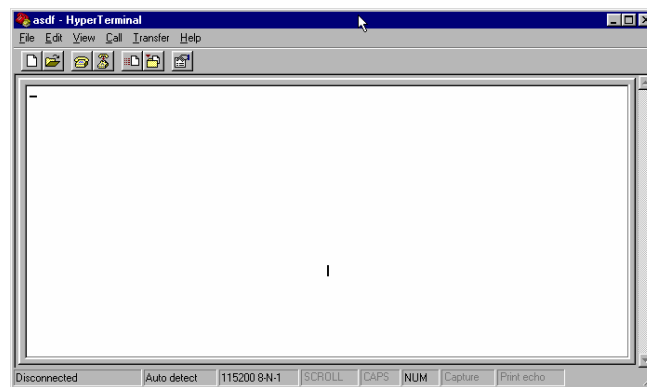
## Connectivity Test

The procedures “Phone/IP Starter Configuration” and “Phonebook Starter Configuration” must be completed before you can do this procedure.

1. These connections must be made:

- MultiVOIP to local phone station  
–OR–  
MultiVOIP to extension of key phone system
- MultiVOIP to command PC
- MultiVOIP to Internet

2. Inbound Phonebook and Outbound Phonebook must both be set up with at least one entry in each. These entries must allow for connection between two voip units.
3. Console messages must be enabled. (If this has not been done already, go, in the MultiVOIP GUI, to Configuration | Logs and select the “Console Messages” checkbox.
4. Make sure that the COM port connection is free so that the HyperTerminal program can use it.
5. Open the **HyperTerminal** program.



6. Use HyperTerminal to receive and record console messages from the MultiVOIP unit. To do so, set up HyperTerminal as follows (setup shown is for Windows NT4; details will differ slightly in other MS operating systems):

- In the upper toolbar of the HyperTerminal screen, click on the **Properties** button.
- In the “Connect To” tab of the **Connection Properties** dialog box, click on the **Configure** button.
- In the next dialog box, on the “General” tab, set “Maximum Speed” to 115200 bps.
- On the “Connection” tab, set connection preferences to:
  - Data bits:** 8
  - Parity:** none
  - Stop bits:** 1
- Click **OK** twice to exit settings dialog boxes.

7. Make VOIP call on a local phone line accessing PSTN directly or through key system..

8. Read console messages recorded on HyperTerminal.

Console Messages from **Originating VOIP**. The voip unit that originates the call will send back messages like that shown below.

```
[00026975] CAS[0] : RX : ABCD = 1, 1, 1, 1,Pstn State[1] TimeStamp : 26975
[00027190] CAS[0] : TX : ABCD = 1, 1, 1, 1
[00027190] PSTN: cas seizure detected on 0
[00027440] CAS[0] : TX : ABCD = 0, 0, 0, 0
[00033290] PSTN:call detected on 0 num=17637175662*
[00033290] SIP[0]:destAddr = TA:200.2.10.5:1720,NAME:Mounds
View,TEL:17637175662,17637175662
[00033290] SIP[0]:srcAddr = NAME:New York,TA:200.2.9.20
[00033440] SIP [0]:cmCallStateProceeding
[00033500] SIP[0]: Remote Information (Q931): MultiVOIP - T1
[00033565] CAS[0] : TX : ABCD = 1, 1, 1, 1
[00033675] SIP [0]: MasterSlaveStatus=Slave
[00033675] SIP[0]:FastStart Setup Not Used
[00033690] CAS[0] : TX : ABCD = 1, 1, 1, 1
[00033755] SIP[0]: Coder used 'g7231'
[00033810] PSTN:pstn call connected on 0
```

Console Messages from **Terminating VOIP**. The voip unit connected to the phone where the call is answered will send back messages like that shown below.

```
[00170860] SIP[0]: New incoming call
[00170860] PSTNIF : Placing call on channel 0 Outbound digit 7175662
[00170885] CAS[0] : TX : ABCD = 1, 1, 1, 1
[00171095] SIP [0]: MasterSlaveStatus=Master
[00171105] CAS[0] : RX : ABCD = 1, 1, 1, 1,Pstn State[7] TimeStamp : 171105
[00171105] SIP[0]: Coder used 'g7231'
[00171110] SIP[0]:FastStart Setup Not Used
[00171110] SIP[0]: Already opened the outgoing logical channel
[00171110] SIP[0]: Coder used 'g7231'
[00171315] CAS[0] : RX : ABCD = 0, 0, 0, 0,Pstn State[9] TimeStamp : 171315
[00172275] PSTN: dialing digit ended on 0
[00172285] PSTN: pstn proceeding indication on 0
[00172995] CAS[0] : RX : ABCD = 1, 1, 1, 1,Pstn State[12] TimeStamp : 172995
[00173660] CAS[0] : TX : ABCD = 1, 1, 1, 1
[00173760] PSTN:pstn call connected on 0
```

9. When you see the following message, end-to-end voip connectivity has been achieved.

**“PSTN: pstn call connected on X”**

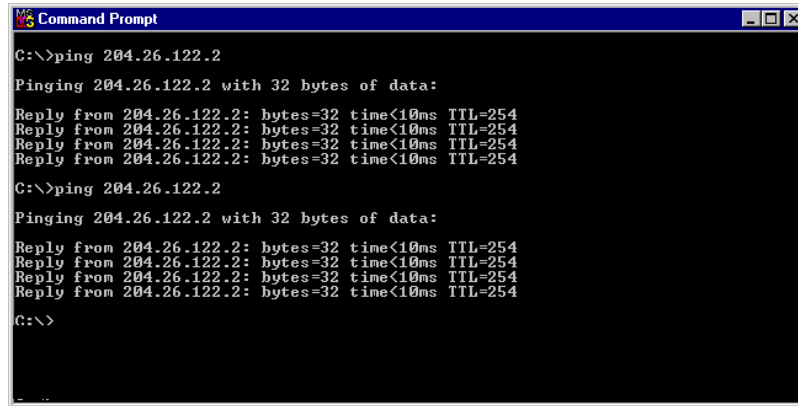
where x is the number of the voip channel carrying the call

10. If the HyperTerminal messages do not confirm connectivity, go to the *Troubleshooting* procedure below.

## Troubleshooting

If you cannot establish connectivity between two voips in the system, follow the steps below to determine the problem.

1. Ping both MultiVOIP units to confirm connectivity to the network.



```
Command Prompt
C:\>ping 204.26.122.2
Pinging 204.26.122.2 with 32 bytes of data:
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
C:\>ping 204.26.122.2
Pinging 204.26.122.2 with 32 bytes of data:
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
C:\>
```

2. Verify the telephone connections.

- Check cabling. Are connections well seated? To correct receptacle?
- Are telephone Interface Parameter settings correct?  
Remember that each voip channel is separately configurable.

3. Verify phonebook configuration.

4. Observe console messages while placing a call. Look for error messages indicating phonebook problems, network problems, voice-coder mismatches, etc.

---

## **Chapter 3: Mechanical Installation and Cabling**

## Introduction

When MVPFX2-4 or MVPFX2-8 units are to be installed into a rack, two able-bodied persons should participate. The MVPFX2-2 is a table-top unit that can generally be handled easily by one person.

Please read the safety notices before beginning installation.

## Safety Warnings

### General Safety

This product must be disconnected from its power source and from the telephone network interface when servicing.

### Lithium Battery Caution

A lithium battery on the voice/fax channel board provides backup power for the timekeeping capability. The battery has an estimated life expectancy of ten years.

When the battery starts to weaken, the date and time may be incorrect. If the battery fails, the board must be sent back to Multi-Tech Systems for battery replacement.

**Warning:** There is danger of explosion if the battery is incorrectly replaced.

### Ethernet (WAN) Ports Caution

**Caution:** The Ethernet ports (often labeled "WAN") are not designed to be connected to a Public Telecommunication Network.

## Safety Warnings Telecom

1. Never install telephone wiring during a lightning storm.
2. Never install a telephone jack in wet locations unless the jack is specifically designed for wet locations.
3. This product is to be used with UL and UL listed computers.
4. Never touch uninsulated telephone wires or terminals unless the telephone line has been disconnected at the network interface.
5. Use caution when installing or modifying telephone lines.
6. Avoid using a telephone (other than a cordless type) during an electrical storm. There may be a remote risk of electrical shock from lightning.
7. Do not use a telephone in the vicinity of a gas leak.
8. To reduce the risk of fire, use only a UL-listed 26 AWG or larger telecommunication line cord.

## Unpacking Your MultiVOIP

When unpacking your MultiVOIP, check to see that all of the items shown are included in the box. For the various MultiVOIP models, the contents of the box will be different. Study the particular illustration below that is appropriate to the model you have purchased. If any box contents are missing, contact MultiTech Tech Support at 1-800-972-2439.

### Unpacking the MVPFX2-4/8

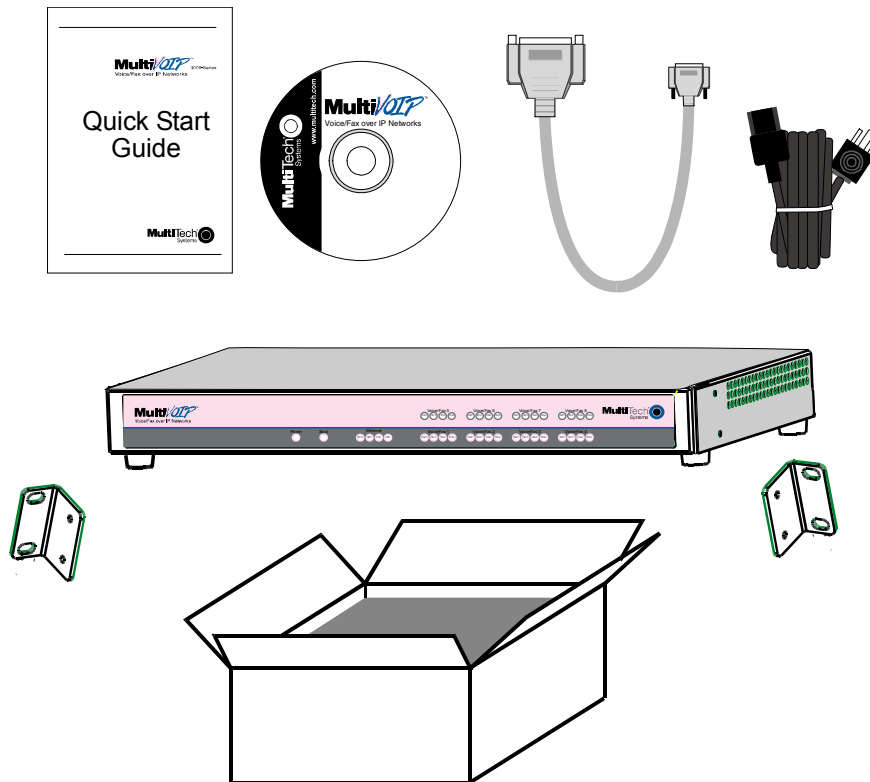


Figure 3-1: Unpacking the MVPFX2-4/8

## Unpacking the MVPFX2-2

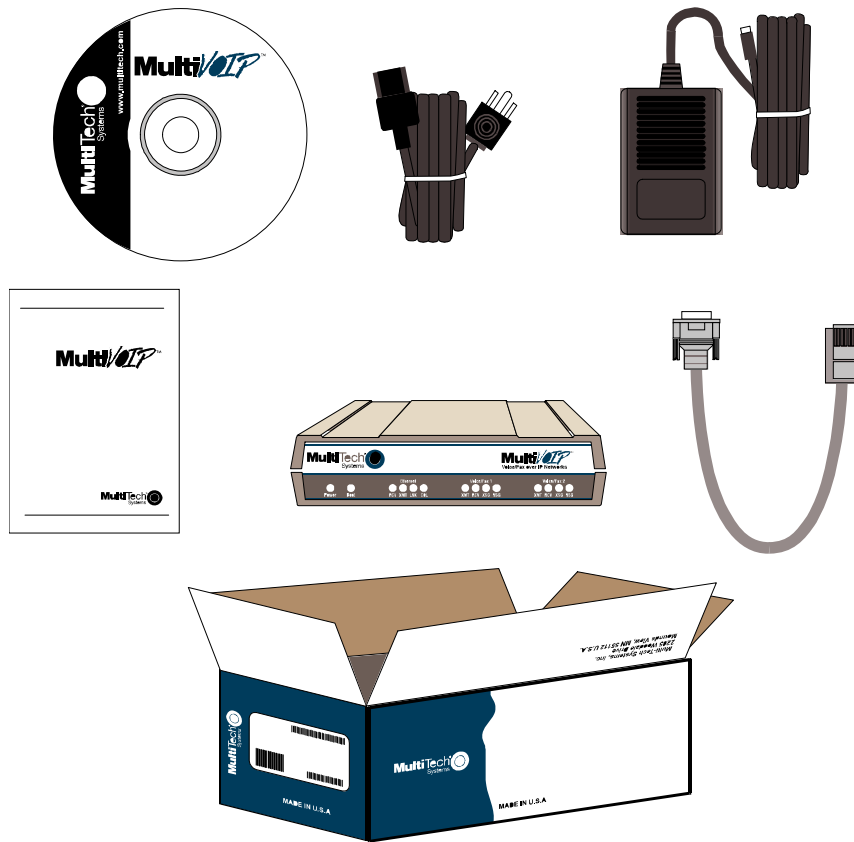
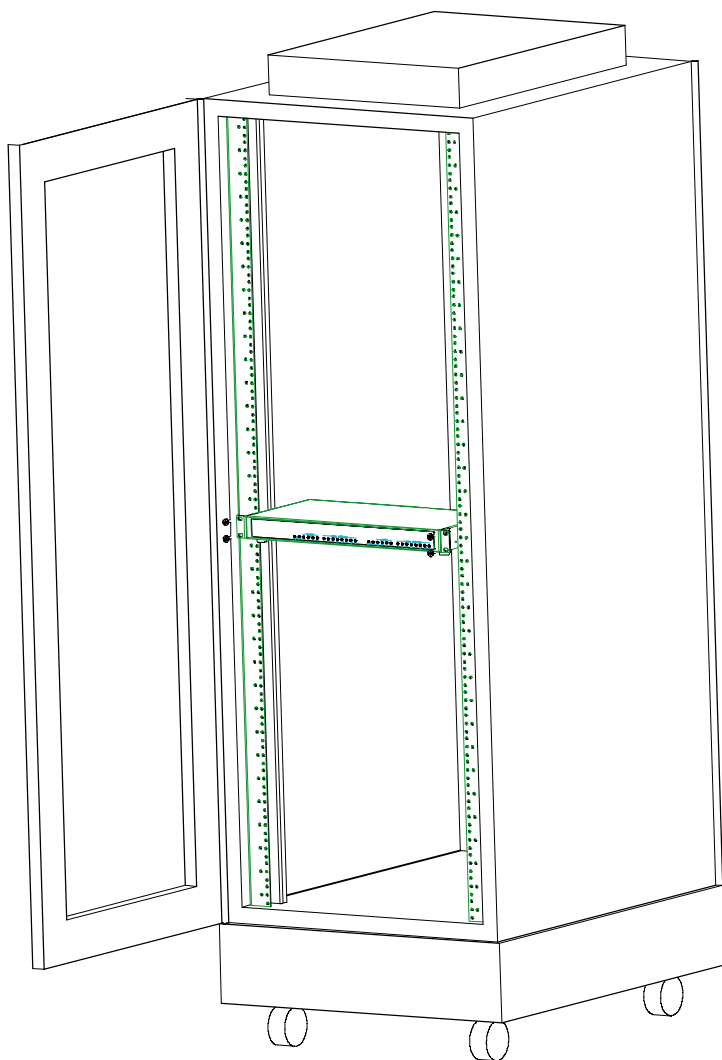


Figure 3-2: Unpacking the MVPFX2-2



### Rack Mounting Instructions for MVPFX2-4 & MVPFX2-8

The MultiVOIPs can be mounted in an industry-standard EIA 19-inch rack enclosure, as shown in Figure 3-3.



**Figure 3-3: Rack-Mounting (MVPFX2-4 or MVPFX2-8)**

## Safety Recommendations for Rack Installations of MVPFX2-4 or MVPFX2-8

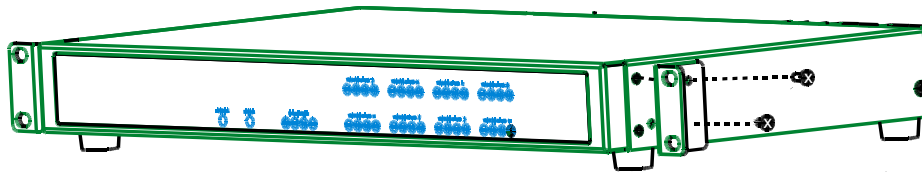
Mounting: Mechanically, this unit is designed for a one-high industry standard EIA 19-inch rack enclosure. The product must be installed by qualified service personnel in a restricted-access area, in accordance with articles 110-16, 10-17, and 110-18 of the National Electrical Code, ANSI/NFPA 70.

- Ensure proper installation of the unit in a closed or multi-unit enclosure by following the recommended installation as defined by the enclosure manufacturer. Do not place the unit directly on top of other equipment or place other equipment directly on top of the unit.
- If installing the unit in a closed or multi-unit enclosure, ensure adequate airflow within the rack so that the maximum recommended ambient temperature is not exceeded.
- Ensure that the unit is properly connected to earth ground by verifying that it is reliably grounded when mounted within a rack. If a power strip is used, ensure that the power strip provides adequate grounding of the attached apparatus.
- When mounting the equipment in the rack, make sure mechanical loading is even to avoid a hazardous condition, such as loading heavy equipment in rack unevenly. The rack used should safely support the combined weight of all the equipment it supports.
- Ensure that the mains supply circuit is capable of handling the load of the equipment. See the power label on the equipment for load requirements (full specifications for MultiVOIP models are presented in chapter 1 of this manual).
- Maximum ambient temperature for the unit is 60 degrees Celsius (140 degrees Fahrenheit) at 20-90% non-condensing relative humidity.
- This equipment should only be installed by properly qualified service personnel.
- Only connect like circuits. In other words, connect SELV (Secondary Extra Low Voltage) circuits to SELV circuits and TN (Telecommunications Network) circuits to TN circuits.
- To reduce the risk of shock, all access doors should be closed during normal operation of the equipment.

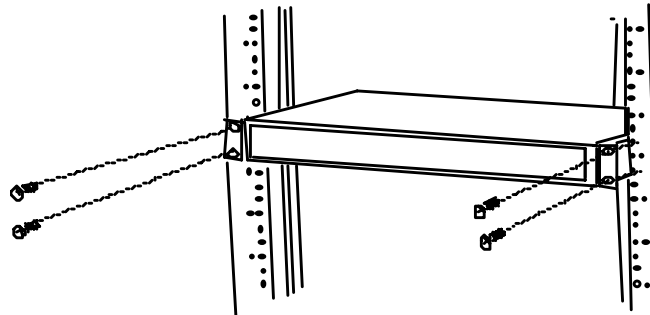
## 19-Inch Rack Enclosure Mounting Procedure

Attaching the MultiVOIP to a rack-rail of an EIA 19-inch rack enclosure will certainly require two persons. Essentially, the technicians must attach the brackets to the MultiVOIP chassis with the screws provided, as shown in Figure 3-4, and then secure unit to rack rails by the brackets, as shown in Figure 3-5. Because equipment racks vary, screws for rack-rail mounting are not provided. Follow the instructions of the rack manufacturer and use screws that fit.

1. Position the right rack-mounting bracket on the MultiVOIP using the two vertical mounting screw holes.
2. Secure the bracket to the MultiVOIP using the two screws provided.
3. Position the left rack-mounting bracket on the MultiVOIP using the two vertical mounting screw holes.
4. Secure the bracket to the MultiVOIP using the two screws provided.
5. Remove feet (4) from the MultiVOIP unit.
6. Mount the MultiVOIP in the rack enclosure per the rack manufacturer's mounting procedure.



**Figure 3-4: Bracket Attachment for Rack Mounting  
(MVPFX2-4 & MVPFX2-8)**

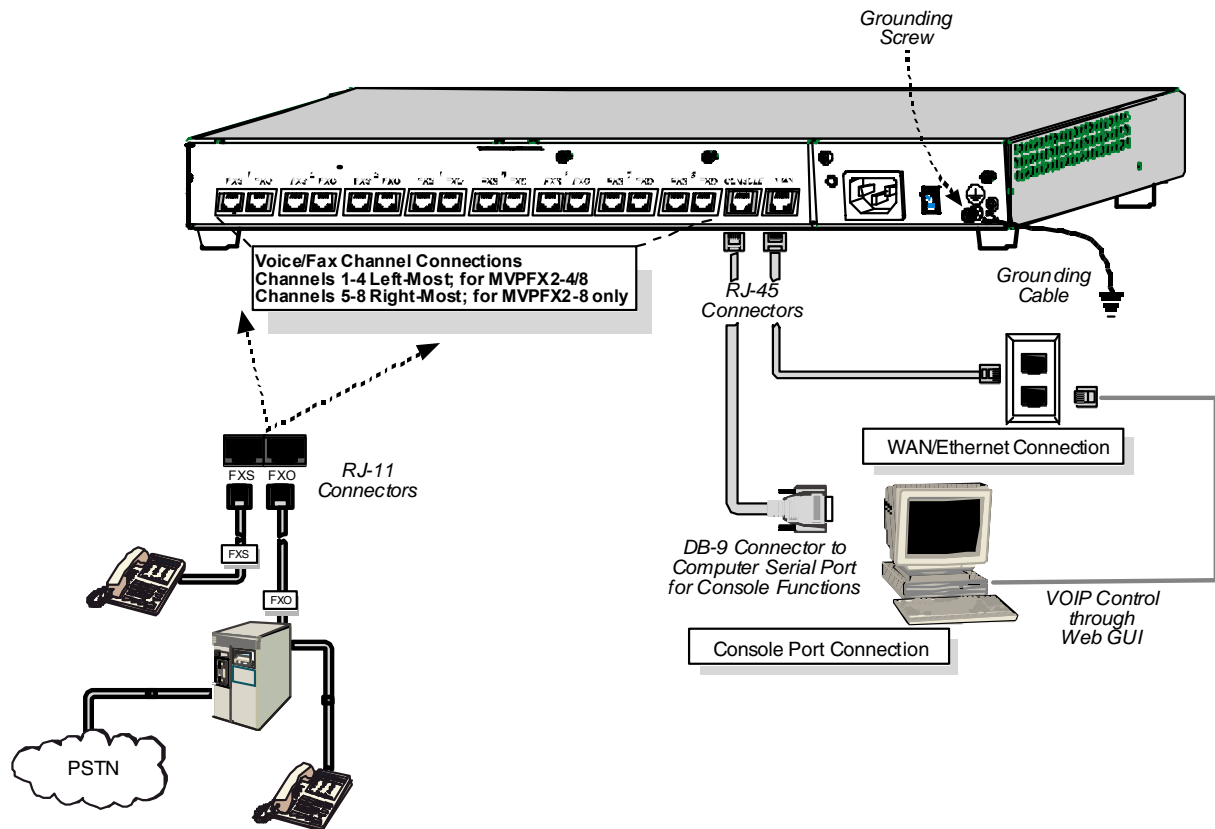


**Figure 3-5: Attaching MultiVOIP to Rack Rail  
(MVPFX2-4 & MVPFX2-8)**

## Cabling Procedure for MVPFX2-4/8

Cabling involves connecting the MultiVOIP to your LAN and telephone equipment.

1. Connect the power cord supplied with your MultiVOIP to a live AC outlet and to the power connector on the back of the MultiVOIP as shown at top right in Figure 3-6.



**Figure 3-6: Cabling for MVPFX-4/8**

2. Connect the MultiVOIP to a PC by using an RJ-45 (male) to DB-9 (female) cable. Plug the RJ-45 end of the cable into the Console port of the MultiVOIP and the other end into the PC serial port. See Figure 3-6.
3. Connect a network cable to the **WAN 10/100** connector on the back of the MultiVOIP. Connect the other end of the cable to your network switch.
4. **For an FXS connection:** *Examples - analog phone, fax machine, Key Telephone System*

Connect one end of an RJ-11 phone cord to the Channel 1 **FXS** connector on the back of the MultiVOIP. Connect the other end to the device.

**For an or FXO connection:** *Examples - PBX extension, POTS line from telco central office*

Connect one end of an RJ-11 phone cord to the Channel 1 **FXO** connector on the back of the MultiVOIP. Connect the other end to the phone jack.

5. Repeat step 4 to connect the remaining telephone equipment to each channel on your MultiVOIP. Although a MultiVOIP's channels are often all configured identically, each channel is individually configurable. So, for example, some channels of a MultiVOIP might use the FXO interface and others the FXS.
6. Ensure that the unit is properly connected to earth ground by verifying that it is reliably grounded when mounted within a rack.
 

This can be accomplished by connecting a grounding wire between the chassis grounding screw (see Figure 3-6) and a metallic object that will provide an electrical ground.
7. Turn on power to the MultiVOIP by placing the ON/OFF switch on the back panel to the ON position. Wait for the **Boot** LED on the MultiVOIP to go off before proceeding. This may take a few minutes.
 

With the connections made, you are ready to contact the web GUI and begin configuring the MultiVOIP.

## Cabling Procedure for MVPFX2-2

Cabling involves connecting the MultiVOIP to your LAN and telephone equipment.

1. Connect the power cord supplied with your MultiVOIP to the power connector on the back of the MultiVOIP and to a live AC outlet as shown in Figure 3-7.

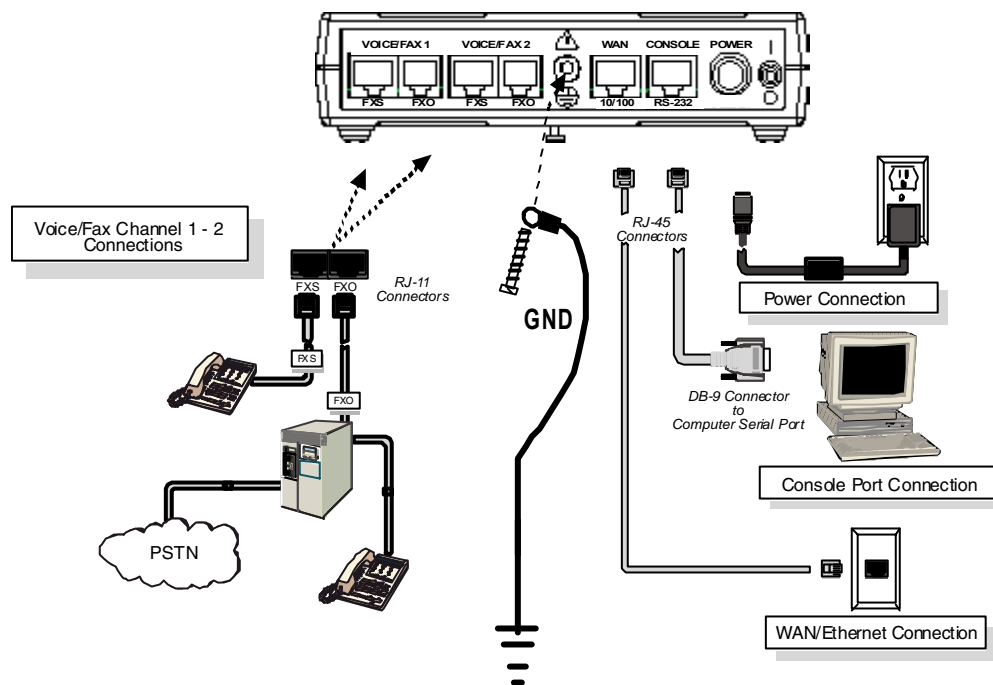


Figure 3-7: Cabling for MVPFX2-2

2. Connect the MultiVOIP to a PC by using a RJ-45 (male) to DB-9 (female) cable. Plug the RJ-45 end of the cable into the Console port of the MultiVOIP and the other end into the PC serial port. See Figure 3-7.
3. Connect a network cable to the **WAN 10/100** connector on the back of the MultiVOIP. Connect the other end of the cable to your network switch.

4. **For an FXS connection:** *Examples - analog phone, fax machine, Key Telephone System*

Connect one end of an RJ-11 phone cord to the Channel **1 FXS** connector on the back of the MultiVOIP. Connect the other end to the device or phone jack.

- For an or FXO connection:** *Examples - PBX extension, POTS line from telco central office*

Connect one end of an RJ-11 phone cord to the Channel **1 FXO** connector on the back of the MultiVOIP. Connect the other end to the device or phone jack.

5. Repeat the above step to connect the remaining telephone equipment to the second channel on your MultiVOIP.
6. Ensure that the unit is properly connected to earth ground by verifying that it is reliably grounded when mounted within a rack.

This can be accomplished by connecting a grounding wire between the chassis and a metallic object that will provide an electrical ground.

7. Turn on power to the MultiVOIP by placing the ON/OFF switch on the back panel to the ON position. Wait for the **BOOT** LED on the MultiVOIP to go off before proceeding. This may take a few minutes.

Proceed to the *MultiVOIP & Auxiliary Software* chapter for considerations about the Java and browser requirements in relation to the MultiVOIP web GUI.

---

## **Chapter 4: MultiVOIP & Auxiliary Software**

## Introduction

The software (firmware) that runs the MVPFX2 unit resides within the MVPFX2 unit and is contacted through a web browser. As such, there is no MultiVOIP configuration software to install. However, the PC operating the web browser GUI must be equipped with an up-to-date version of Java. If an up-to-date version is not already present on the PC, it must be installed from the MultiVOIP CD or from the Java website.

The Java software, the other auxiliary software, and the User Guide are contained on the MultiVOIP product CD. Because the CD is auto-detectable, it will start up automatically when you insert it into your CD-ROM drive. When you have finished loading the Java program, you can view and print the User Guide by clicking on the **View Manuals** icon.

Java is necessary to operate the MultiVOIP GUI. In addition to Java, other optional 3rd-party software packages are necessary to take advantage of certain optional auxiliary MultiVOIP functions. These include programs for SysLog, FTP, and TFTP.

## Summary

Configuring software for your MultiVOIP entails three tasks:

- (1) loading an up-to-date version of Java onto the PC to enable the web-GUI to operate and, if required, loading other auxiliary software,
- (2) setting values for telephony and IP parameters that will fit your system (this is "Technical Configuration" and it is discussed in Chapter 5), and
- (3) establishing "phonebooks" that contain the various dialing patterns for VOIP calls made to different locations (this is "Phonebook Configuration" and it is discussed in Chapter 6).



---

## **Chapter 5: Technical Configuration**

## Configuring the MultiVOIP

There are two ways in which the MultiVOIP must be configured before operation: technical configuration and phonebook configuration.

**Technical Configuration.** First, the MultiVOIP must be configured to operate with technical parameter settings that will match the equipment with which it interfaces. There are five types of technical parameters that must be set.

These technical parameters pertain to

- (1) its operation in an IP network,
- (2) its operation with telephony equipment,
- (3) its transmission of voice and fax messages,
- (4) certain telephony attributes that are common to particular nations or regions,
- (5) selecting the method by which log reports will be made accessible.

The process of specifying values for the various parameters in these seven categories is what we call “technical configuration” and it is described in this chapter.

**Phonebook Configuration.** The second type of configuration that is required for the MultiVOIP pertains to the phone number dialing sequences that it will receive and transmit when handling calls. Dialing patterns will be affected by both the PBX/telephony equipment and the other VOIP devices that the MultiVOIP unit interacts with. We call this “Phonebook Configuration,” and, for analog MultiVOIP units, it is described in Chapter 6. The *Quick Start Guide* chapter presents additional information on phonebook setup.

**Local/Remote Configuration.** The MultiVOIP is configured through a web browser. The MultiVOIP is factory configured to this IP address: **192.168.2.1**.

Certain functions (like update/upgrade of firmware version) and changing the MultiVOIP’s IP address can be done locally via a hard-wired connection between a PC serial port and the MultiVOIP’s “Console” port.

## Configuration by Web GUI

This manual primarily describes configuration of the MultiVOIP with the web GUI.

### Pre-Requisites




To complete the configuration of the MultiVOIP unit, you **must** know several things about the overall system.

Before configuring your MultiVOIP Gateway unit, you must know the values for several IP and telephone parameters that describe the IP network system and telephony system (PBX or telco central office equipment) with which the digital MultiVOIP will interact. A summary of this configuration information appears on page 60 (“Config Info CheckList”).

### IP Parameters


The following parameters must be known about the network (LAN, WAN, Internet, etc.) to which the MultiVOIP will connect:

➔	<i>Ask your computer network administrator.</i>	<i>Info needed to operate: all MultiVOIP models.</i>
		IP Network Parameters: Record for each VOIP Site in System
	<ul style="list-style-type: none"> <li>• IP Address</li> </ul>	
	<ul style="list-style-type: none"> <li>• IP Mask</li> </ul>	
	<ul style="list-style-type: none"> <li>• Gateway</li> </ul>	
	<ul style="list-style-type: none"> <li>• Domain Name Server (DNS) Info</li> </ul>	

Write down the values for these IP parameters. You will need to enter these values in the “IP Parameters” screen in the Configuration section of the MultiVOIP software. You must have this IP information about *every* VOIP in the system.

### Telephony Interface Parameters

The following parameters must be known about the PBX or telco central office equipment to which the analog MultiVOIP will connect:

➔	<b>Phone Parameters</b>	
	<i>Ask phone company or telecom manager.</i>	
	 Telephony Interface Parameters: Record for this VOIP Site	
	<ul style="list-style-type: none"> <li>Which interface type is to be used? FXS_____ FXO_____</li> </ul>	
	<ul style="list-style-type: none"> <li>If FXS, determine whether the line will be used for a phone, fax, or KTS (key telephone system)</li> </ul>	
	<ul style="list-style-type: none"> <li>If FXO, determine if line will be an analog PBX extension or an analog line from a telco central office</li> </ul>	

### Config Info CheckList

Type of Configuration Info Gathered	MultiVOIP Configuration screen on which to enter the Info	Info Obtained √	Info Entered √
IP Info for voip unit <ul style="list-style-type: none"> <li>IP address</li> <li>Gateway</li> <li>DNS IP (if used)</li> </ul>	IP Parameters		
Interface Type (Choices: FXS, FXO)  *In FXO/FXS systems, channels used for phone, fax, or key system are FXS; channels used for analog PBX extensions or analog telco lines are FXO.	Interface Parameters.		
Country Code	Regional Parameters		
<b>Reminder:</b> Be sure to <b>Save &amp; Apply</b> after entering configuration values.			

## Procedure for Configuration by Web GUI (Summary)

After the MultiVOIP configuration software has been installed in the 'Command' PC (which is connected to the MultiVOIP unit), several steps must be taken to configure the MultiVOIP to function in its specific setting. Although the summary below includes all of these steps, some are optional.

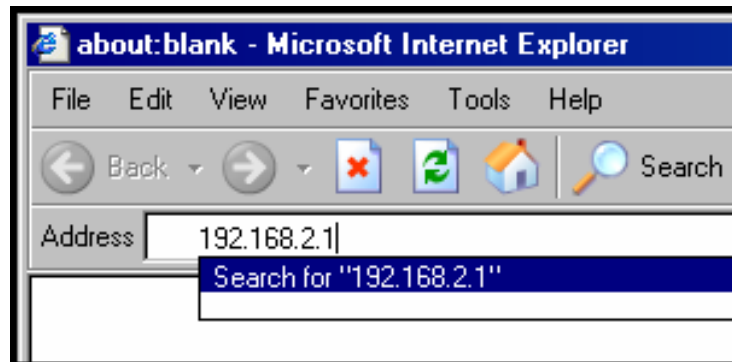
1. Check Power and Cabling.
2. Start MultiVOIP web-based Configuration Program.
3. Confirm Connection.
4. Familiarize yourself with configuration parameter screens and how to access them.
5. Set IP Parameters.
6. Set Voice/Fax Parameters.
7. Set Telephony Interface Parameters.
8. Set SIP Call Signaling parameters.
9. Set Regional Parameters (Phone Signaling Tones & Cadences and setup for built-in Remote Configuration/Command Modem).
10. Set Log Reporting Method (GUI, locally in MultiVOIP Configuration program; or SMTP, via email).
11. View System Info screen and set updating interval (optional).
12. Set Packetization Time.
13. Save the MultiVOIP configuration.

When technical configuration is complete, you will need to configure the MultiVOIP's inbound and outbound phonebooks in the *Phonebook Configuration* chapter.

## Local Configuration Procedure (Detailed)

You can begin the configuration process after assuring that an up-to-date browser and up-to-date Java application are present on your computer. You can establish your configuration or modify it at any time by contacting the web-GUI through a browser.

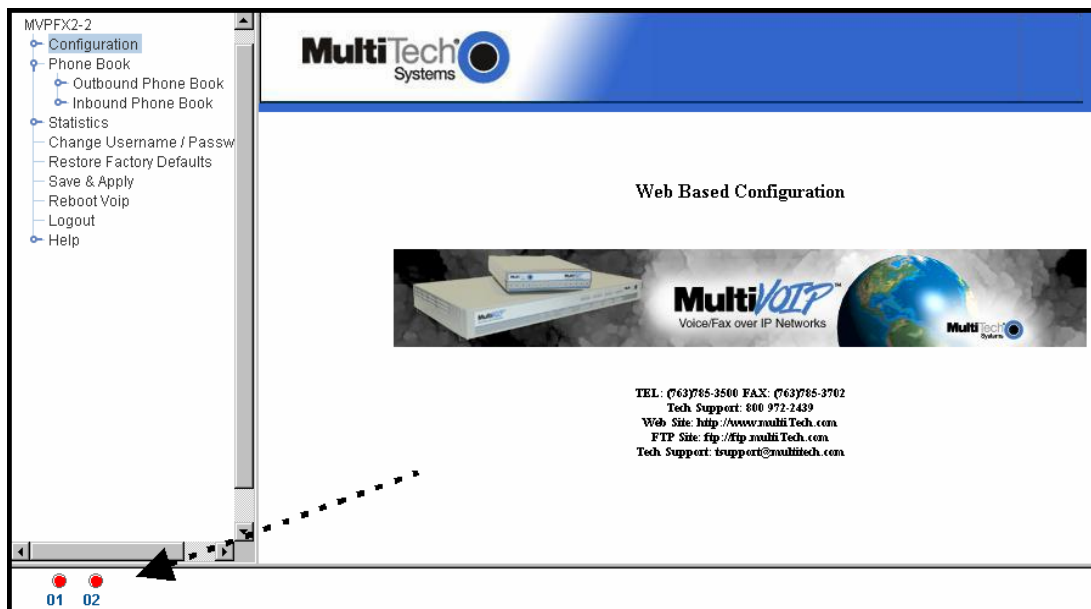
1. **Check Power and Cabling.** Be sure the MultiVOIP is turned on and connected to the computer via the WAN/Ethernet connection (this entails RJ-45 cabling between MultiVOIP and the WAN/Ethernet network).
2. **Start MultiVOIP Configuration Program.** Launch the MultiVOIP program from a web browser. The default IP address assigned to the MultiVOIP is 192.168.2.1. However, this address can be changed by connecting the PC to the MultiVOIP's **Console** port and using a communications program



Operation of MultiVOIP through web GUI requires up-to-date version of Java. If Java has not yet been installed, follow these instructions:

- A. Install up-to-date Java program from MultiVOIP product CD (on first use only).
- B. Open web browser.
- C. Browse to IP address of MultiVOIP unit.
- D. If username and password have been established, enter them when when prompted.
- E. Set browser to allow pop-ups. The MultiVOIP Web GUI makes extensive use of pop-up windows to access screens and commands.

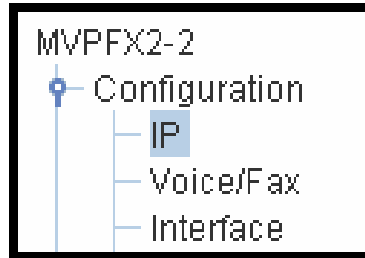
3. **Confirm Connection.** When the PC is in communication with the MultiVOIP through the web browser, you will see an icon for each voip channel in the lower left corner of the screen. The icon is green when the channel is in use and red when idle.



4. **Configuration Parameter Groups: Getting Familiar, Learning About Access.** The first part of configuration concerns IP parameters, Voice/FAX parameters, Telephony Interface parameters, Call Signaling parameters, Regional parameters, Logs/Traces, System Information, and Packetization Time. In the MultiVOIP software, these eight types of parameters are grouped together under “Configuration” and each has its own dialog box for entering and viewing values.

To access the dialog box for these parameter groups, click on the name of the parameter group in the sidebar menu of the browser.

5. **Set IP Parameters.** Click on **IP Parameters** in the sidebar list.



In each field, enter the values that fit your particular network.

Current Permission: Read/Write

### IP Parameters

**Diffserv Parameters**

Call Control PHB

VoIP Media PHB

Type II

**IP Parameters**

Gateway Name

Enable DHCP

IP Address

IP Mask

Gateway

**DNS**

Enable DNS

DNS Server IP Address

**FTP Server**

Enable

OK

Cancel

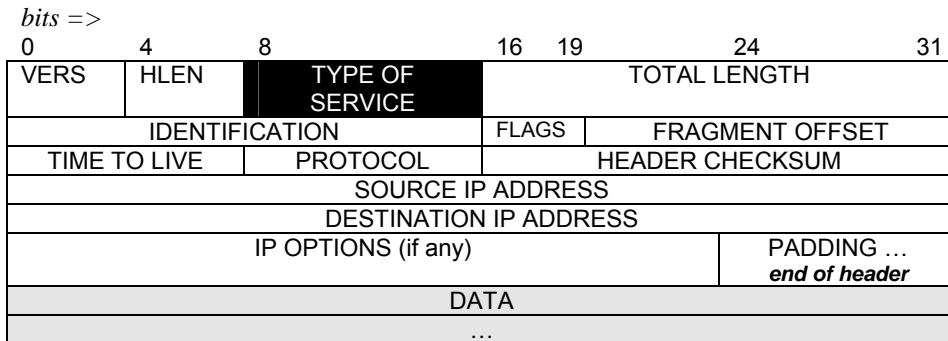
The **IP Parameters** fields are described in the tables and text passages below. Note that both DiffServ parameters (Call Control PHB and VoIP Media PHB) must be set to zero if you enable Packet Prioritization (802.1p). Nonzero DiffServ values negate the prioritization scheme.

<b>IP Parameter Definitions</b>		
<b>Field Name</b>	<b>Values</b>	<b>Description</b>
<b>Ethernet Parameters</b>		
Frame Type	Type II, SNAP	Must be set to match network's frame type. Default is Type II.
<b>DiffServ Parameter fields</b>	<p>DiffServ PHB (Per Hop Behavior) values pertain to a differential prioritizing system for IP packets as handled by DiffServ-compatible routers. There are 64 values, each with an elaborate technical description. These descriptions are found in TCP/IP standards RFC2474, RFC2597, and, for present purposes, in RFC3246, which describes the value 34 (34 decimal; 22 hex) for Assured Forwarding behavior (default for Call Control PHB) and the value 46 (46 decimal; 2E hexadecimal) for Expedited Forwarding behavior (default for Voip Media PHB). Before using values other than these default values of 34 and 46, consult these standards documents and/or a qualified IP telecommunications engineer.</p> <p>To disable DiffServ, configure both fields to 0 decimal.</p> <p>The passage following this table explains DiffServ in the context of the IP datagram.</p>	
Call Control PHB	0 - 63 default = 34 .	Value is used to prioritize call setup IP packets.
Voip Media PHB	0 - 63 default = 46 <i>n</i>	Value is used to prioritize the RTP/RTCP audio IP packets.



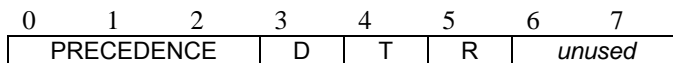
<b>IP Parameter Definitions (cont'd)</b>		
<b>Field Name</b>	<b>Values</b>	<b>Description</b>
<b>IP Parameter fields</b>		
Gateway Name	alphanumeric	Descriptor of current voip unit to distinguish it from other units in system.
Enable DHCP	Y/N disabled by default	Dynamic Host Configuration Protocol is a method for assigning IP address and other IP parameters to computers on the IP network in a single message with great flexibility. IP addresses can be static or temporary depending on the needs of the computer.
IP Address	4-places, 0-255	The unique LAN IP address assigned to the MultiVOIP.
IP Mask	4-places, 0-255	Subnetwork address that allows for sharing of IP addresses within a LAN.
Gateway	4-places, 0-255.	The IP address of the device that connects your MultiVOIP to the Internet.

**The IP Datagram with Header, Its Type-of-Service field, & DiffServ**



The TOS field consists of eight bits, of which only the first six are used. These six bits are called the “Differentiated Service Codepoint” or DSCP bits.

**The Type of Service or “TOS” field**



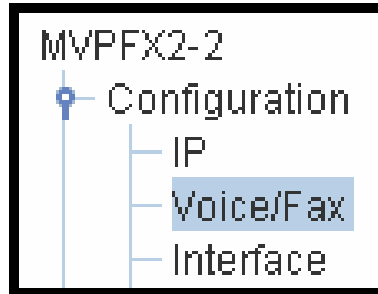
three precedence have eight values, 0-7, ranging from “normal” precedence (value of 0) to “network control” (value of 7). When set, the *D* bit requests low delay, the *T* bit requests high throughput, and the *R* bit requests high reliability.

Routers that support DiffServ can examine the six DSCP bits and prioritize the packet based on the DSCP value. The DiffServ Parameters fields in the MultiVOIP IP Parameters screen allow you to configure the DSCP bits to values supported by the router. Specifically, the Voip Media PHB field relates to the prioritizing of audio packets (RTP and RTCP packets) and the Call Control PHB field relates to the prioritizing of non-audio packets (packets concerning call set-up and tear-down, gatekeeper registration, etc.).

The MultiVOIP Call Control PHB parameter defaults to 34 decimal (22 hex; 100010 binary – consider vis-à-vis TOS field above) for Assured Forwarding behavior. The MultiVOIP Voip Media PHB parameter defaults to the value 46 decimal (2E hex; 101110 binary – consider vis-à-vis TOS field above). To disable DiffServ, configure both fields to 0 decimal.

IP Parameter Definitions (cont'd)		
Field Name	Values	Description
<b>DNS Parameter fields</b>		
Enable DNS	Y/N Default = disabled	Enables Domain Name Space/System function where computer names are resolved using a worldwide distributed database.
DNS Server IP Address	4-places, 0-255.	IP address of specific DNS server to be used to resolve Internet computer names.
<b>FTP Parameter fields</b>		
FTP Server Enable	Y/N Default = enabled See "FTP Server File Transfers" in <i>Operation &amp; Maintenance</i> chapter.	MultiVOIP unit has an FTP Server function so that firmware and other important operating software files can be transferred to the voip via the network.

8. Set Voice/FAX Parameters. Click on Voice/FAX in the sidebar list.



In each field, enter the values that fit your particular network.

Current Permission: Read/Write

### Voice/Fax Parameters

Select Channel : Channel 01

**Voice Gain**

Input 0 dB    Output 0 dB

**DTMF**

Duration 60 ms

DTMF: Out of Band

Out Of Band Mode Rfc2833

**Coder**

Manual     Automatic

Selected Coder G.723.1 audio

Max bandwidth 10 kbps

**Auto Call**

Auto Call None

Phone Number

**Dynamic Jitter Buffer**

Minimum Jitter Value 20 ms

Maximum Jitter Value 200 ms

Initial Jitter Value 20

**Automatic Disconnection**

Jitter Value 250 ms     Consecutive Packets Lost 250

Call Duration 300 secs     Network Disconnection 300 secs

**Fax**

Fax Enable

Max Baud Rate 7200 kbps

Fax Volume -16.5 dB

Jitter Value 1 ms

Mode FRF 11

**Advanced Features**

Silence Compression

Echo Cancellation

Forward Error Correction

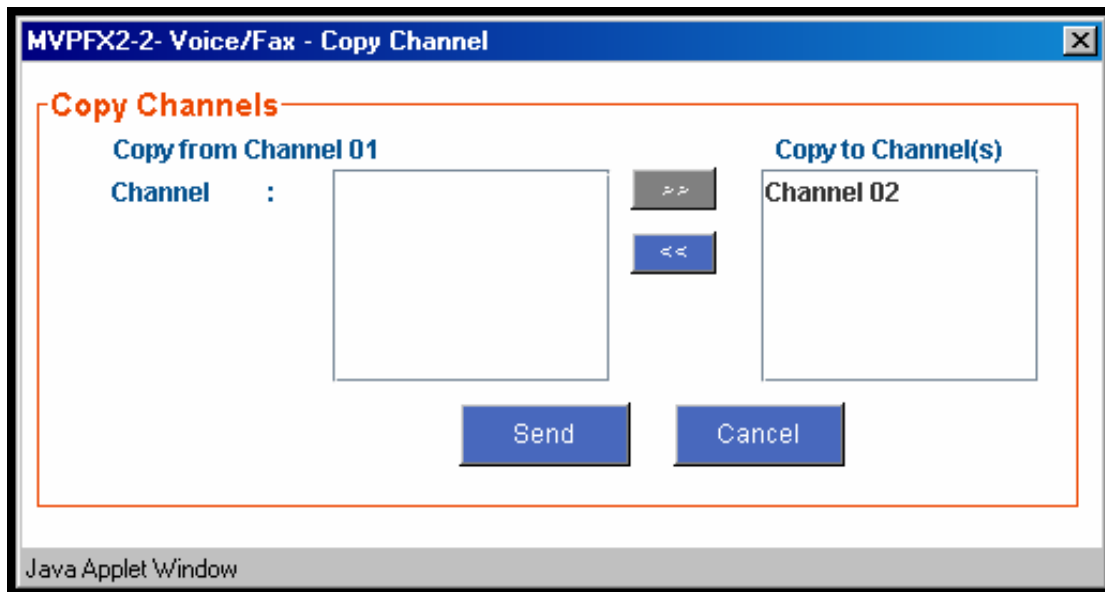
OK

Cancel

Copy Channel

Default

Note that Voice/FAX parameters are applied on a channel-by-channel basis. However, once you have established a set of Voice/FAX parameters for a particular channel, you can apply this entire set of Voice/FAX parameters to other channels by using the **Copy Channel** button and its dialog box.



The **Voice/FAX Parameters** fields are described in the tables below.

Voice/Fax Parameter Definitions		
Field Name	Values	Description
Default	--	When this button is clicked, all Voice/FAX parameters are set to their default values.
Select Channel	1-2 (FX2-2) 1-4 (FX2-4) 1-8 (FX2-8)	Channel to be configured is selected here.
Copy Channel	--	Copies the Voice/FAX attributes of one channel to another channel. Attributes can be copied to multiple channels or all channels at once.
Voice Gain	--	Signal amplification (or attenuation) in dB.
Input Gain	+31dB to -31dB	Modifies audio level entering voice channel before it is sent over the network to the remote VOIP. The default & recommended value is <b>0 dB</b> .
Output Gain	+31dB to -31dB	Modifies audio level being output to the device attached to the voice channel. The default and recommended value is <b>0 dB</b> .

Voice/Fax Parameter Definitions (cont'd)		
Field Name	Values	Description
<b>DTMF Parameters</b>		
Duration (DTMF)	60 - 3000 ms	When <b>DTMF: Out of Band</b> is selected, this setting determines how long each DTMF digit 'sounds' or is held. Default = 100 ms.
DTMF In/Out of Band	Out of Band, or Inband	When <b>DTMF Out of Band</b> is selected, the MultiVOIP detects DTMF tones at its input and regenerates them at its output. When <b>DTMF Inband</b> is selected, the DTMF digits are passed through the MultiVOIP unit as they are received.
Out of Band Mode	RFC 2833, SIP Info	<p><b>RFC2833 method.</b> Uses an RTP mode defined in RFC 2833 to transmit the DTMF digits.</p> <p><b>SIP Info method.</b> Generates dual tone multi frequency (DTMF) tones on the telephony call leg. The SIP INFO message is sent along the signaling path of the call.</p> <p>You must set this parameter per the capabilities of the remote endpoint with which the voip will communicate. The RFC2833 method is the more common of the two methods.</p>

Voice/Fax Parameter Definitions (cont'd)		
Coder Parameters		
Coder	Manual or Automatic	Determines whether selection of coder is manual or automatic. When Automatic is selected, the local and remote voice channels will negotiate the voice coder to be used by selecting the highest bandwidth coder supported by both sides without exceeding the Max Bandwidth setting. G.723, G.729, or G.711 are negotiated.
Selected Coder	G.711 a/u law 64 kbps; G.726, @ 32 kbps; G.723.1 @ 6.3 kbps; G.729, 8kbps;	Select from a range of coders with specific bandwidths. The higher the bps rate, the more bandwidth is used. The channel that you are calling must have the same voice coder selected.  Default = G.723.1 @ 6.3 kbps. Here 64K of digital voice are compressed to 6.3K, allowing several simultaneous conversations over the same bandwidth that would otherwise carry only one.  To make selections from the Selected Coder drop-down list, the Manual option must be enabled.
Max bandwidth (coder)	11 - 128 kbps	This field lets you specify the maximum bandwidth allowed for this channel.  If coder is to be selected automatically ("Auto" setting), then enter a value for maximum bandwidth.

Voice/Fax Parameter Definitions		
Field Name	Values	Description
<b>AutoCall Parameters</b>		
Auto Call	AutoCall	<p>The <b>AutoCall</b> option enables the local MultiVOIP to call a remote MultiVOIP without the user having to dial a Phone Directory Database number. As soon as you access the local MultiVOIP voice/fax channel, the MultiVOIP immediately connects to the remote MultiVOIP identified in the <b>Phone Number</b> box of this option.</p> <p>This function applies on a channel-by-channel basis. It would not be appropriate for this function to be applied to a channel that serves in a pool of available channels for general phone traffic. This function requires an entry in the Outgoing phonebook of the local MultiVOIP and a matched setting in the Inbound Phonebook of the remote voip.</p>
Phone Number	--	<p>Phone number used for Auto Call function or Offhook Alert Timer function. This phone number must correspond to an entry in the Outbound Phonebook of the local MultiVOIP and in the Inbound Phonebook of the remote MultiVOIP (unless a gatekeeper unit is used in the voip system).</p>



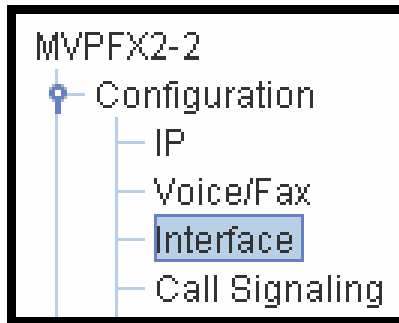
Voice/Fax Parameter Definitions (cont'd )		
Field Name	Values	Description
<b>Dynamic Jitter</b>		
Dynamic Jitter Buffer		<b>Dynamic Jitter</b> defines a minimum and a maximum jitter value for voice communications. When receiving voice packets from a remote MultiVOIP, varying delays between packets may occur due to network traffic problems. This is called Jitter. To compensate, the MultiVOIP uses a Dynamic Jitter Buffer. The Jitter Buffer enables the MultiVOIP to wait for delayed voice packets by automatically adjusting the length of the Jitter Buffer between configurable minimum and maximum values. The length of the jitter buffer directly effects the voice delay between MultiVOIP gateways.
Minimum Jitter Value	60 to 400 ms	The minimum dynamic jitter buffer of <b>60</b> milliseconds is the minimum delay that would be acceptable over a low jitter network. Default = 150 msec

Voice/Fax Parameter Definitions (cont'd)		
Field Name	Values	Description
<b>Dynamic Jitter</b>		
Maximum Jitter Value	20 to 200 ms	The maximum dynamic jitter buffer of <b>200</b> milliseconds is the maximum delay tolerable over a high jitter network. Default = 200 msec
Initial Jitter Value	20 - 200 ms default = 20 ms	The starting value (in ms) of the Jitter Buffer. This value will change itself from the starting value depending on the needs of the jitter buffer. For example, if you set the initial value at 100, it may end up scaling itself down to 20 or up to 200 to meet the needs of operating conditions.

Voice/Fax Parameter Definitions (cont'd )		
Field Name	Values	Description
<b>Auto Disconnect</b>		
Automatic Disconnection	--	The <b>Automatic Disconnection</b> group provides four options which can be used singly or in any combination.
Jitter Value	1-65535 milliseconds	The <b>Jitter Value</b> defines the average inter-arrival packet deviation (in milliseconds) before the call is automatically disconnected. The default is <b>300</b> milliseconds. A higher value means voice transmission will be more accepting of jitter. A lower value is less tolerant of jitter. Inactive by default. When active, default = 300 ms. However, value must equal or exceed Dynamic Minimum Jitter Value.
Call Duration	1-65535 seconds	<b>Call Duration</b> defines the maximum length of time (in seconds) that a call remains connected before the call is automatically disconnected. Inactive by default. When active, default = <b>180</b> sec. This may be too short for most configurations, requiring upward adjustment.
Consecutive Packets Lost	1-65535	<b>Consecutive Packets Lost</b> defines the number of consecutive packets that are lost after which the call is automatically disconnected. Inactive by default. When active, default = <b>30</b>
Network Disconnection	1 to 65535 seconds; Default = 300 sec.	Specifies how long to wait before disconnecting the call when IP network connectivity with the remote site has been lost.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	Values	Description
<b>Advanced Features</b>		
Silence Compression	Y/N	Determines whether silence compression is enabled (checked) for this voice channel.  With <b>Silence Compression</b> enabled, the MultiVOIP will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth that is being used by the voice channel. Default = on.
Echo Cancellation	Y/N	Determines whether echo cancellation is enabled (checked) for this voice channel.  <b>Echo Cancellation</b> removes echo and improves sound quality. Default = on.

7. Set Telephony Interface Parameters. Click on **Interface** in the sidebar list.



In each field, enter the values that fit your particular network.

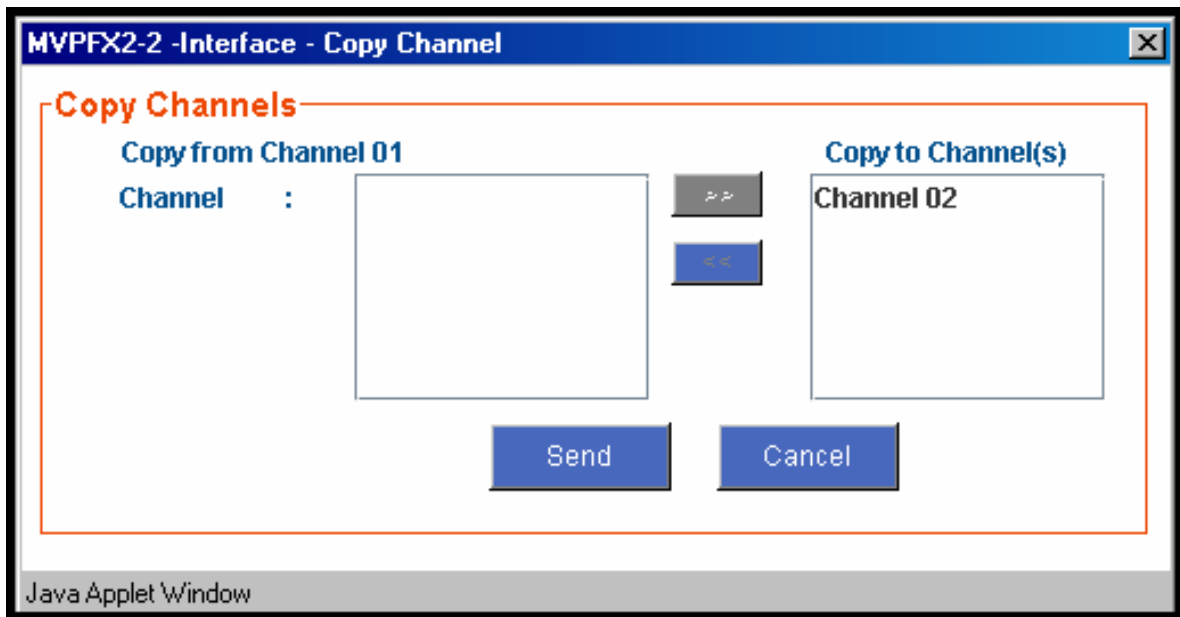
A screenshot of the 'Interface Parameters' configuration window. At the top, it shows 'Select Channel: Channel 01' and 'Interface Type: FXS (Loop Start)'. The window is divided into several sections:

- FXS Options:** Ring Count (8), Current Loss (checkbox), Flash Detection Range (Min: 100 ms, Max: 1000 ms).
- Caller ID:** Type: BellCore, Enable (checkbox).
- FXO Options:** Ring Count (2), No Response Timer (200 secs), Flash Generation (600 ms).
- Disconnect Options:** Current Loss (checkbox), Current Loss Timer (500 ms), DTMF Tone (checkbox), Sequence (dropdown), Silence Detection (checkbox), Type (One Way), Timer (0 secs).
- Dialing Options:** Regeneration (Pulse, DTMF), Inter Digit Timer (2 secs), Inter Digit Regeneration Timer (100 ms).

On the right side, there are four buttons: OK, Cancel, Copy Channel, and Default.

The kinds of parameters for which values must be chosen depend on which type of telephony supervisory signaling or interface is used (FXO or FXS Loop Start). We present here the various parameters grouped and organized by interface type.

Note that Interface parameters are applied on a channel-by-channel basis. However, once you have established a set of Interface parameters for a particular channel, you can apply this entire set of Interface parameters to other channels by using the **Copy Channel** button and its dialog box.



**FXS Loop Start Parameters.** The parameters applicable to FXS Loop Start are shown in the figure below and described in the table that follows.

FXS Loop Start Interface: Parameter Definitions		
Field Name	Values	Description
Select Channel	1-2 (MVPFX2-2); 1-4 (MVPFX2-4); 1-8 (MVPFX2-8)	Indicates the voip channel to which parameter values will be assigned.
Interface Type	FXS Loop Start	The value of this field determines whether this channel uses the FXS Loop Start interface type or the FXO interface type. We are here discussing the FXS Loop Start option.

<b>FXS Loop Start Interface: Parameter Definitions (cont'd)</b>		
<b>Field Name</b>	<b>Values</b>	<b>Description</b>
<b>FXS Options fields</b>		
Ring Count , FXS	1-99	Maximum number of rings that the MultiVOIP will issue before giving up the attempted call.
Current Loss	Y/N	When enabled, the MultiVOIP will interrupt loop current in the FXS circuit to initiate a disconnection. This tells the device connected to the FXS port to hang up. The Multi-VOIP cannot drop the call; the FXS device must go on hook.
<b>Flash Detection Range fields</b>		
Min/Max	<i>for Min. and Max.,</i> 50 - 1500 milliseconds	For a received flash hook to be regarded as such by the MultiVOIP, its duration must fall between the minimum and maximum values given here.
<b>Caller ID fields</b>		
Type	Bellcore	The MultiVOIP currently supports only one implementation of Caller ID. That implementation is Bellcore type 1 with Caller ID placed between the first and second rings of the call.
Enable	Y/N	Caller ID information is a description of the remote calling party received by the called party. The description has three parts: name of caller, phone number of caller, and time of call. The 'time-of-call' portion is always generated by the receiving MultiVOIP unit (on FXS channel) based on its date and time setup.



FXS Loop Start Interface: Parameter Definitions (cont'd)		
Field Name	Values	Description
<b>Caller ID fields</b>		
Enable (cont'd)	Y/N	In general, the forms of the 'Caller Name' and 'Caller Phone Number' differ depending on the IP transmission protocol used (H.323, SIP, or SPP) and upon entries in the phonebook screens of the remote (CID generating) voip unit. For MVPFX2 units, only the SIP-related Caller ID options are available. The CID Name and Number appearing on the phone at the terminating FXS end will come either from a central office switch (showing a PSTN phone number), or the phonebook of the remote (CID sending) voip unit.
<b>Dialing Options fields</b>		
Regeneration	Pulse, DTMF	Indicates which type of dialing must be regenerated, either pulses or DTMF. For MVPFX2 units, DTMF is always used.
Inter Digit Timer	1 - 10 seconds	This is the length of time that the MultiVOIP will wait between digits. When the time expires, the MultiVOIP will look in the outbound phonebook for the number entered and place the call accordingly. Default = 2.
Inter Digit Regeneration Timer	in milliseconds	The length of time between the outputting of DTMF digits. Default = 100 ms.

The Caller ID feature has dependencies on both the telco central office and the MultiVOIP phone book. See the diagram series after the FXO Parameters section below.

**FXO Parameters.** The parameters applicable to the FXO telephony interface type are shown in the figure below and described in the table that follows.

Current Permission: Read/Write

### Interface Parameters

Select Channel: Channel 01 ▼
Interface Type: FXO ▼

#### FXS Options

#### FXO Options

Ring Count 2

No Response Timer 200 secs

Flash Generation 600 ms

#### Disconnect Options

Current Loss    Current Loss Timer 500 ms

DTMF Tone    Sequence \* ▼ + 0 ▼

Silence Detection    Type One Way ▼    Timer 25 secs

#### Caller ID

Type: BellCore ▼

Enable

#### Dialing Options

#### Regeneration

Pulse     DTMF

Inter Digit Timer 2 secs

Inter Digit Regeneration Timer 100 ms

OK  
Cancel  
Copy Channel  
Default

<b>FXO Interface: Parameter Definitions*</b>		
<b>Field Name</b>	<b>Values</b>	<b>Description</b>
Select Channel	1-2 (MVPFX2-2); 1-4 (MVPFX2-4); 1-8 (MVPFX2-8)	Indicates the voip channel to which parameter values will be assigned.
Interface Type	FXO	Enables FXO functionality.
<b>Caller ID fields</b>		
Type	Bellcore	The MultiVOIP currently supports only one implementation of Caller ID. That implementation is Bellcore type 1 with Caller ID placed between the first and second rings of the call.
Enable	Y/N	<p>Caller ID information is a description of the remote calling party received by the called party. The description has three parts: name of caller, phone number of caller, and time of call. The 'time-of-call' portion is always generated by the receiving MultiVOIP unit (on FXS channel) based on its date and time setup.</p> <p>In general, the forms of the 'Caller Name' and 'Caller Phone Number' differ depending on the IP transmission protocol used and upon entries in the phonebook screens of the remote (CID generating) voip unit. For MVPFX2 units, only the SIP-related Caller ID options are available. The CID Name and Number appearing on the phone at the terminating FXS end will come either from a central office switch (showing a PSTN phone number), or the phonebook of the remote (CID sending) voip unit.</p>

FXO Interface: Parameter Definitions*																																
FXO Options																																
Ring Count	1-99	Number of rings required before the MultiVOIP answers the incoming call.																														
No Response Timer	1 - 65535 (in seconds)	Length of time before call connection attempt is abandoned.																														
Flash Generation	50 - 1500 milliseconds	Length of flash hook that will be generated and sent out when the remote end initiates a flash hook and it is regenerated locally. Default = 600 ms.																														
Disconnect Options																																
Current Loss	Y/N	Disconnection to be triggered by loss of current. That is, when Current Loss is enabled ("Y"), the MultiVOIP will hang up the call at a specified interval after it detects a loss of current initiated by the attached device.																														
Current Loss Timer	200 to 2000 (in milliseconds)	Determines the interval after detection of current loss at which the call will be disconnected.																														
DTMF Tone	Y/N	Enables supervision of call disconnection using DTMF tones.																														
	<table border="1" style="width: 100%; text-align: center;"> <thead> <tr> <th colspan="4">DTMF Tone Pairs</th> <th>Low Tones</th> </tr> </thead> <tbody> <tr> <td></td> <td>1</td> <td>2</td> <td>3</td> <td>A 697Hz</td> </tr> <tr> <td></td> <td>4</td> <td>5</td> <td>6</td> <td>B 770Hz</td> </tr> <tr> <td></td> <td>7</td> <td>8</td> <td>9</td> <td>C 852Hz</td> </tr> <tr> <td></td> <td>*</td> <td>0</td> <td>#</td> <td>D 941Hz</td> </tr> <tr> <td>High Tones</td> <td>1209Hz</td> <td>1336Hz</td> <td>1447Hz</td> <td>1633Hz</td> </tr> </tbody> </table>		DTMF Tone Pairs				Low Tones		1	2	3	A 697Hz		4	5	6	B 770Hz		7	8	9	C 852Hz		*	0	#	D 941Hz	High Tones	1209Hz	1336Hz	1447Hz	1633Hz
DTMF Tone Pairs				Low Tones																												
	1	2	3	A 697Hz																												
	4	5	6	B 770Hz																												
	7	8	9	C 852Hz																												
	*	0	#	D 941Hz																												
High Tones	1209Hz	1336Hz	1447Hz	1633Hz																												

<b>FXO Interface: Parameter Definitions*</b>		
<b>Disconnect Options</b>		
Sequence	1 <sup>st</sup> tone pair + 2 <sup>nd</sup> tone pair	These are DTMF tone pairs constitute 2-part sequences that signal call disconnection.  Values for first tone pair are: *, #, 0, 1-9, and A-D.  Values for second tone pair are: none, 0, 1-9, A-D, *, and #. The tone pairs 1-9, 0, *, and # are the standard DTMF pairs found on phone sets. The tone pairs A-D are "extended DTMF" tones, which are used for various PBX functions.
Silence Detection	Y/N	Enables/disables silence-detection method of supervising call disconnection.
Type (of Silence Detection)	One-Way or Two-Way	Disconnection to be triggered by silence in one direction only or in both directions simultaneously.
Timer (for Silence Detection)	integer values, in seconds	Duration of silence required to trigger disconnection.
<b>Dialing Options</b>		
Regeneration	Pulse, DTMF	Determines whether digits generated and sent out will be pulse tones or DTMF.
Inter Digit Timer	1 to 10 seconds	This is the length of time that the MultiVOIP will wait between digits. When the time expires, the MultiVOIP will look in the phonebook for the number entered.  Default = 2.
Inter Digit Regeneration Time	50 to 20,000 milliseconds	The length of time between the outputting of DTMF digits.  Default = 100 ms.

The Caller ID feature has dependencies on both the telco central office and the MultiVOIP phone book. See the diagram series below.

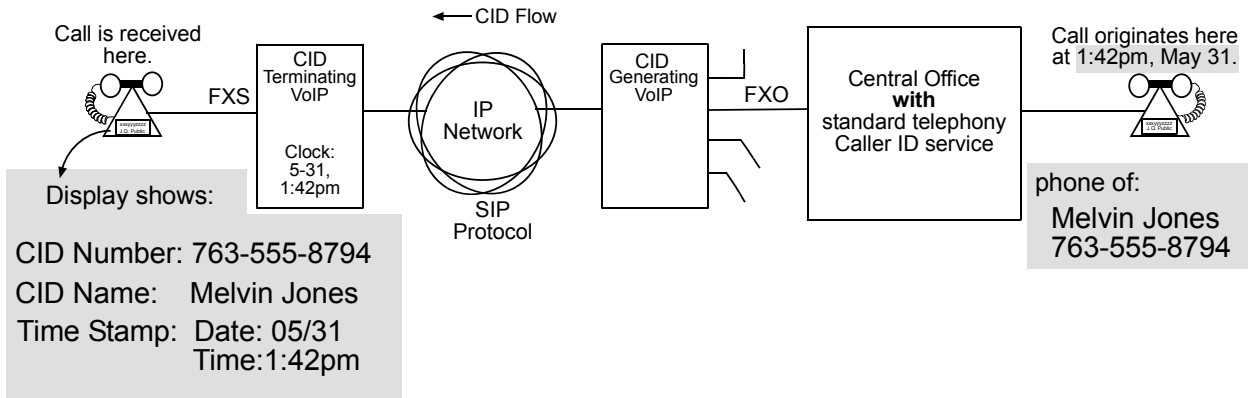


Figure 5-1: Voip Caller ID Case #1 – Call, through telco central office *with* standard CID, enters voip system

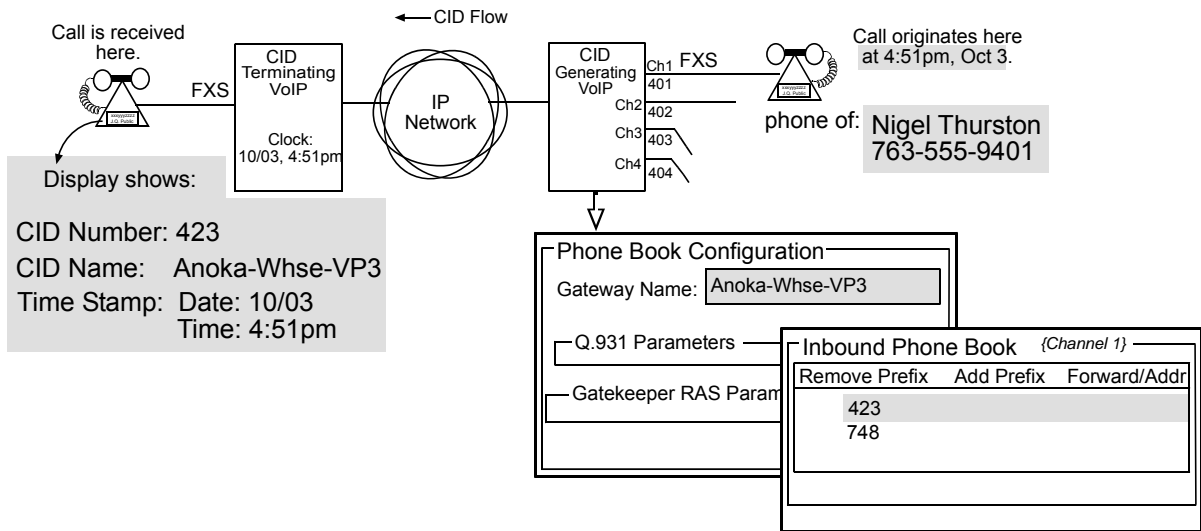
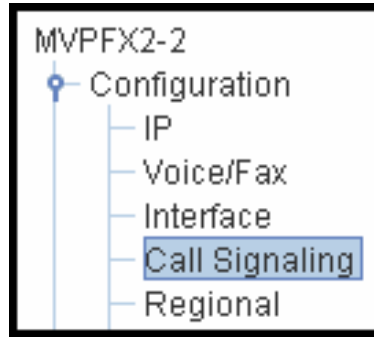


Figure 5-2: Voip Caller ID Case #2 – Remote FXS call on SIP voip system

8. **Set Call Signaling Parameters.** Click on **Call Signaling** in the sidebar list.



Current Permission: Read/Write

### Call Signalling Configuration

**SIP Parameters**

Signaling Port

Use SIP Proxy

**Proxy Parameters**

Proxy Domain Name / IP Address

Append SIP Proxy Domain Name in User ID

Port Number

UserName

Password

Re-RegistrationTime  secs

OK

Cancel

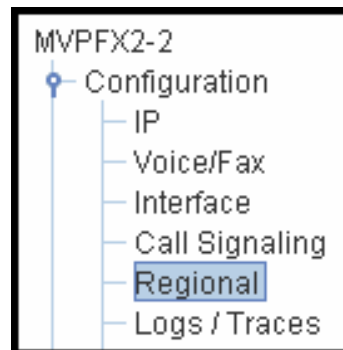
The tables below describe all fields in the **Call Signaling** screen.

<b>Call Signaling Parameter Definitions</b>		
<b>Field Name</b>	<b>Values</b>	<b>Description</b>
SIP Parameters		
Signaling Port	numeric	Port number on which the MultiVOIP UserAgent software module will be waiting for any incoming SIP requests. Default = 5060
Use SIP Proxy	Y/N	Allows the MultiVOIP to work in conjunction with a proxy server.
Proxy Parameters		
Proxy Domain Name / IP Address	n.n.n.n where n=0-255	Network address of the proxy server that the voip is using.
Append SIP Proxy Domain Name in User ID	Y/N	When checked, the domain name of the SIP Proxy serving the MultiVOIP gateway will be included as part of the User ID for that gateway. If unchecked, the SIP Proxy's IP address will be included as part of the User ID instead of the SIP Proxy's domain name.
Port Number		Logical port number for proxy communications.

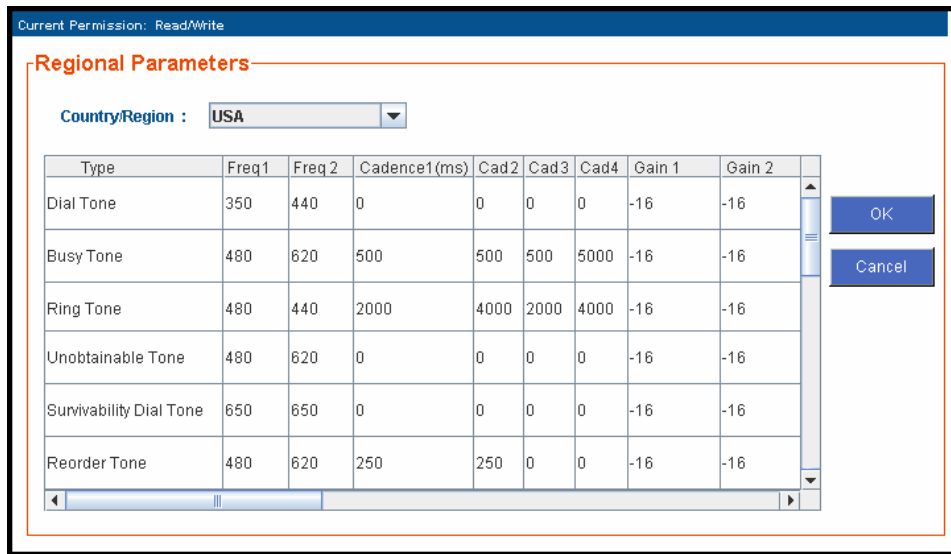


Call Signaling Parameter Definitions		
Field Name	Values	Description
Proxy Parameters (cont'd)		
User Name	Values: alphanumeric	Description: Identifier used when proxy server is used in network. If a proxy server is used in a SIP voip network, all clients must enter both a User Name and a Password before being allowed to make a call.
Password	Values: alphanumeric	Description: Password for proxy server function. See "User Name" description above.
Re-Registration Time	Values: numeric (in seconds)	Description: This is the timeout interval for registration of the MultiVOIP with a SIP proxy server. The time interval begins the moment the MultiVOIP gateway registers with the SIP proxy server and ends at the time specified by the user in the Re-Registration Time field (this field). When/if registration lapses, call traffic routed to/from the MultiVOIP through the SIP proxy server will cease. However, calls in progress will continue to function until they end.

9. **Set Regional Parameters** (Phone Signaling Tones & Cadences).  
Click on **Regional** in the sidebar list.



The **Regional Parameters** screen will appear. For the country selected, the standard set of frequency pairs will be listed for dial tone, busy tone, ‘unobtainable’ tone (fast busy or trunk busy), ring tone, and other, more specialized tones.



In the **Country/Region** field, select the option that fits your particular system. When you choose a Country, you choose an entire set of tones and those tones cannot be altered. To create a nonstandard set of tones, use the “Custom” option.

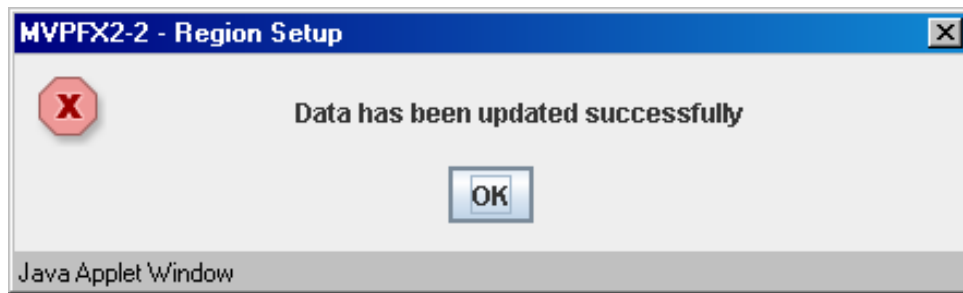
The **Regional Parameters** fields are described in the table below.

“Regional Parameter” Definitions		
Field Name	Values	Description
Country/ Region	Australia, Central America, Chile, Europe, France, Japan, UK, USA, Custom	Name of a country or region that uses a certain set of tone pairs for <b>dial tone</b> , <b>ring tone</b> , <b>busy tone</b> , <b>unobtainable tone</b> (fast busy tone), and <b>re-order tone</b> (a tone pattern indicating the need for the user to hang up the phone).  In some cases, the tone-pair scheme denoted by a country name may also be used outside of that country. The “Custom” option (button) assures that any tone-pairing scheme worldwide can be accommodated.

"Regional Parameter" Definitions		
Field Name	Values	Description
Standard Tones fields		
Type column	dial tone, ring tone, busy tone, unobtainable tone (fast busy), survivability tone, re-order tone	Type of telephony tone-pair for which frequency, gain, and cadence are being presented.
Frequency 1	freq. in Hertz	Lower frequency of pair.
Frequency 2	freq. in Hertz	Higher frequency of pair.

"Regional Parameter" Definitions (cont'd)		
Field Name	Values	Description
Standard Tones fields (cont'd)		
Cadence (msec) On/Off	n/n/n/n four integer time values in milli-seconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote phone ringing, phone busy, connection unobtainable (fast busy), dial tone ("0" indicates continuous tone), survivability, and re-order. Default values differ for different countries/regions. Although most cadences have only two parts (an "on" duration and an "off" duration), some telephony cadences have four parts. Most cadences, then, are expressed as two iterations of a two-part sequence. Although this is redundant, it is necessary to allow for expression of 4-part cadences.
Gain 1	gain in dB +3dB to -31dB and "mute" setting	Amplification factor of lower frequency of pair. This applies to the dial, ring, busy and 'unobtainable' tones that the MultiVOIP outputs as audio to the FXS or FX0 port. <b>Default: -16dB</b>
Gain 2	gain in dB +3dB to -31dB and "mute" setting	Amplification factor of higher frequency of pair. This applies to the dial, ring, busy, and 'unobtainable' (fast busy) tones that the MultiVOIP outputs as audio to the FXS or FX0 port. <b>Default: -16dB</b>

After selecting the appropriate Country/Region for your system, click **OK**. A screen will appear confirming that the configuration has been updated.

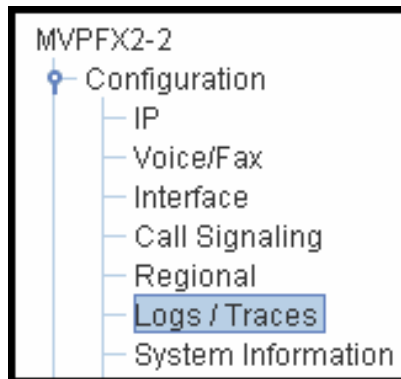


You must select **Save and Apply** in the sidebar menu to make the change permanent.

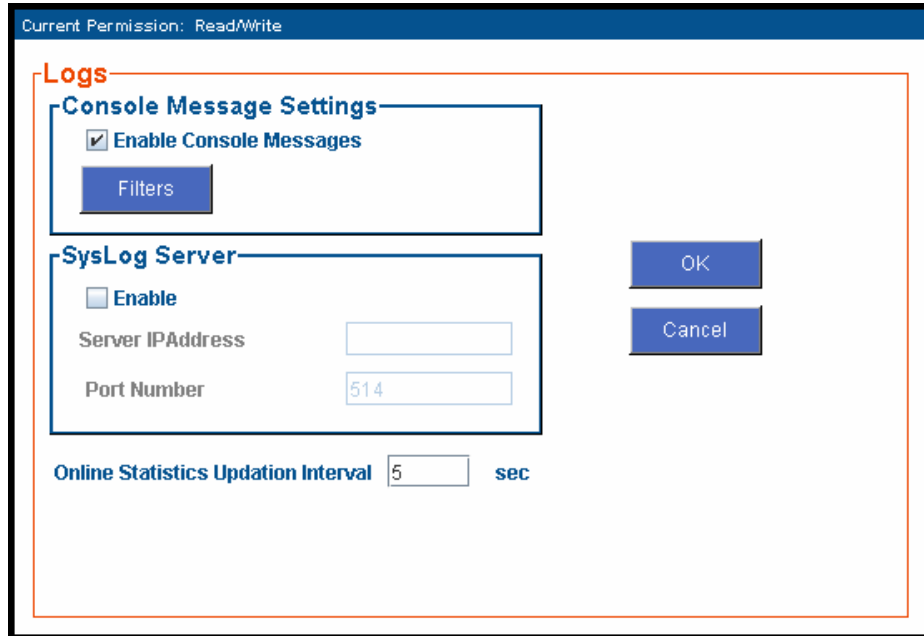
10. **Set Log Reporting Method.** The **Logs** screen lets you choose how the VoIP administrator will receive log reports about the MultiVOIP’s performance and the phone call traffic that is passing through it. Log reports can be received in one of two ways:

- A. as Console Messages accessible through a telecommunications program like HyperTerminal,  
or
- B. through a SysLog Server program.

Click **Logs/Traces** on the sidebar menu to access the Logs/Traces screen.



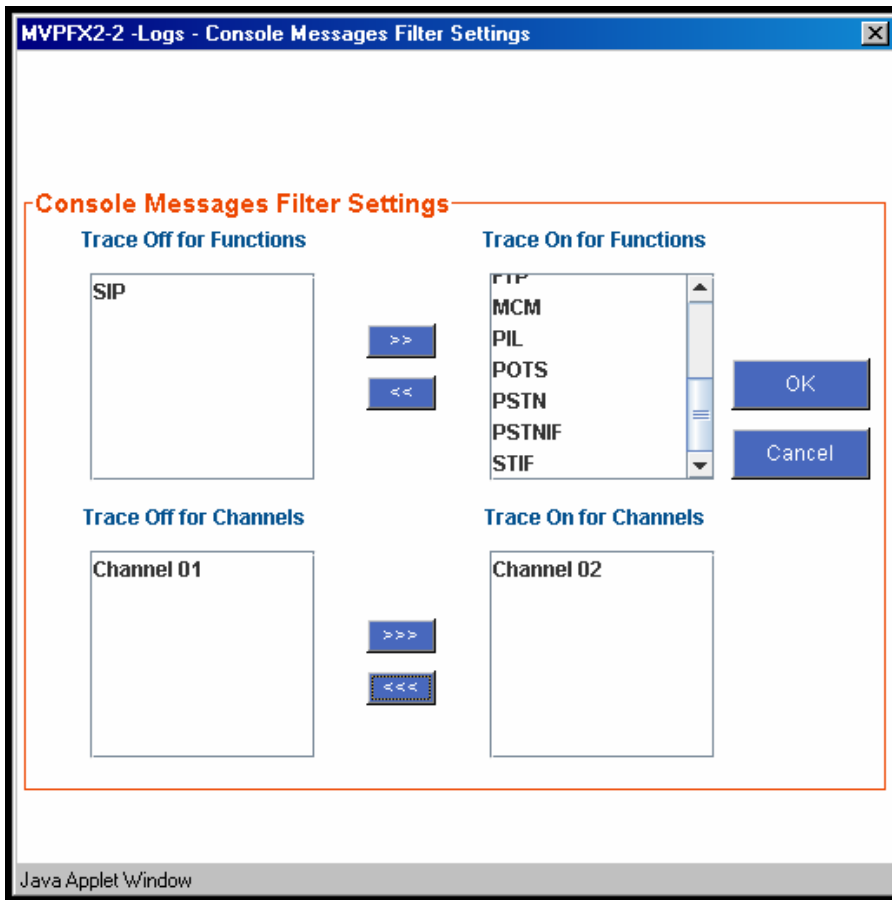
The Logs/Traces screen will appear.



If you enable console messages, you can customize the types of messages to be included/excluded in log reports by clicking on the "Filters" button and using the **Console Messages Filter Settings** screen (see subsequent page). If you use the logging function, select the logging option that applies to your VoIP system design. If you intend to use a SysLog Server program for logging, click in that Enable check box. The common SysLog logical port number is 514.

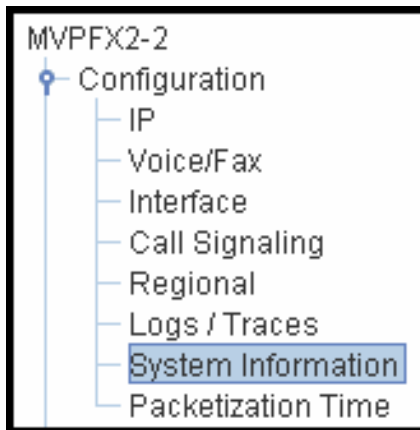
<b>“Logs” Screen Definitions</b>		
<b>Field Name</b>	<b>Values</b>	<b>Description</b>
Enable Console Messages	Y/N	Allows MultiVOIP debugging messages to be read via a basic terminal program like HyperTerminal™ or equivalent. Normally, this should be disabled because it uses MultiVOIP processing resources. Console messages are meant for tech support personnel.
Filters (button)		Click to access secondary screen on where console messages can be included/excluded by category and on a per-channel basis. (See the Console Messages Filter Settings screen on subsequent page.)
SysLog Server Enable	Y/N	This box must be checked if logging is to be done in conjunction with a SysLog Server program. For more on SysLog Server, see <i>Operation &amp; Maintenance</i> chapter.
IP Address	n.n.n.n for n= 0-255	IP address of computer, connected to voip network, on which SysLog Server program is running.
Port	514	Logical port for SysLog Server. 514 is commonly used.
Online Statistics Update Interval	integer	Set the interval (in seconds) at which logging information will be updated.

To customize console messages by category and/or by channel, click on “Filters” and use the **Console Messages Filters Settings** screen.



11. View **System Information** screen and set updating interval (optional).

To reach this dialog box, click **System Information** in the sidebar menu.



This screen presents vital system information at a glance. Its primary use is in troubleshooting.

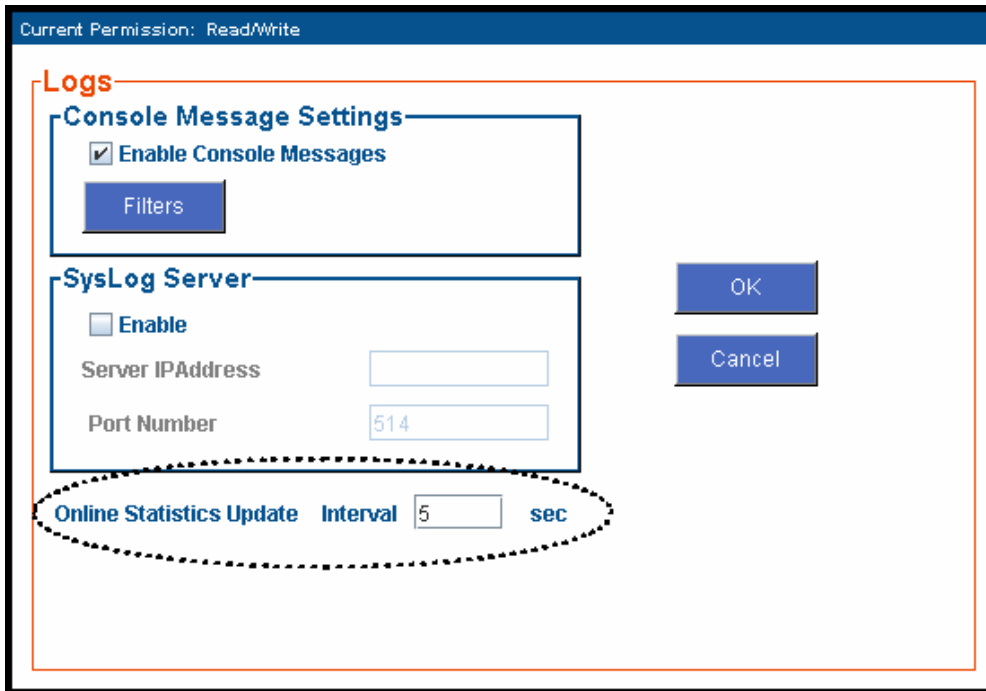


System Information Parameter Definitions		
Field Name	Values	Description
Boot Version	nn.nn	Indicates the version of the code that is used at the startup (booting) of the voip. The boot code version is independent of the software version.
Firmware Version	alpha-numeric	Indicates version of MultiVOIP firmware.



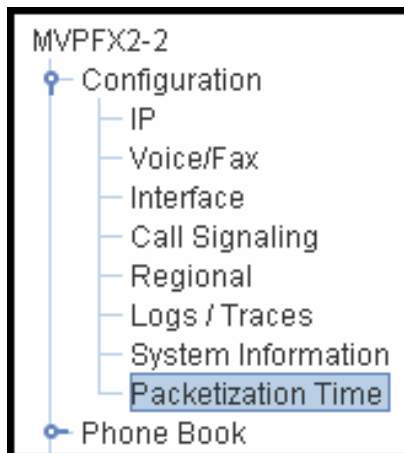
<b>System Information Parameter Definitions (cont'd)</b>		
<b>Field Name</b>	<b>Values</b>	<b>Description</b>
Configur- ation Version	nn.nn.nn. nn alpha- numeric	Indicates version of MultiVOIP Configuration software (which includes screens for IP Parameters, SMTP Parameters, Regional Parameters, etc.).
Phone Book Version	numeric	Indicates the version of the inbound and outbound phonebook portion of the MultiVOIP software.
MSP Version	nn.nn alpha- numeric	Version of DSP (digital signal processor) software used in MultiVOIP.
Kernel Version	Vn_nn_ n.n	Linux kernel version used in MultiVOIP.
Mac Address	alpha- numeric	Denotes the number assigned as the voip unit's unique Ethernet address.
Up Time	days: hours: mm:ss	Indicates how long the voip has been running since its last booting.

The frequency with which several administrative screens are updated (the System Information, Call Progress, and IP Statistics screens) is determined by a setting in the Logs/Traces screen.



12. About Packetization Time.

You can use the **Packetization Time** screen to specify definite packetization rates for coders selected in the Voice/FAX Parameters screen (in the "Coder" group of fields). The Packetization Time screen is accessible in the sidebar list of the main voip software screen. In dealing with RTP parameters, the Packetization Time screen is closely related to both Voice/FAX Parameters and to IP Statistics.



## Packetization Time Screen

Current Permission: Read/Write

**Packetization Time**

Select Channel : Channel 01 ▼

**Packetization Rate (msec per packet)**

G711 A-law	<span style="border: 1px solid gray; padding: 2px;">10</span> ▼	G.726-32 audio	<span style="border: 1px solid gray; padding: 2px;">10</span> ▼
G711 U-law	<span style="border: 1px solid gray; padding: 2px;">10</span> ▼	G.729A audio	<span style="border: 1px solid gray; padding: 2px;">20</span> ▼
G.723.1 audio	<span style="border: 1px solid gray; padding: 2px;">60</span> ▼		

OK

Cancel

Copy Channel

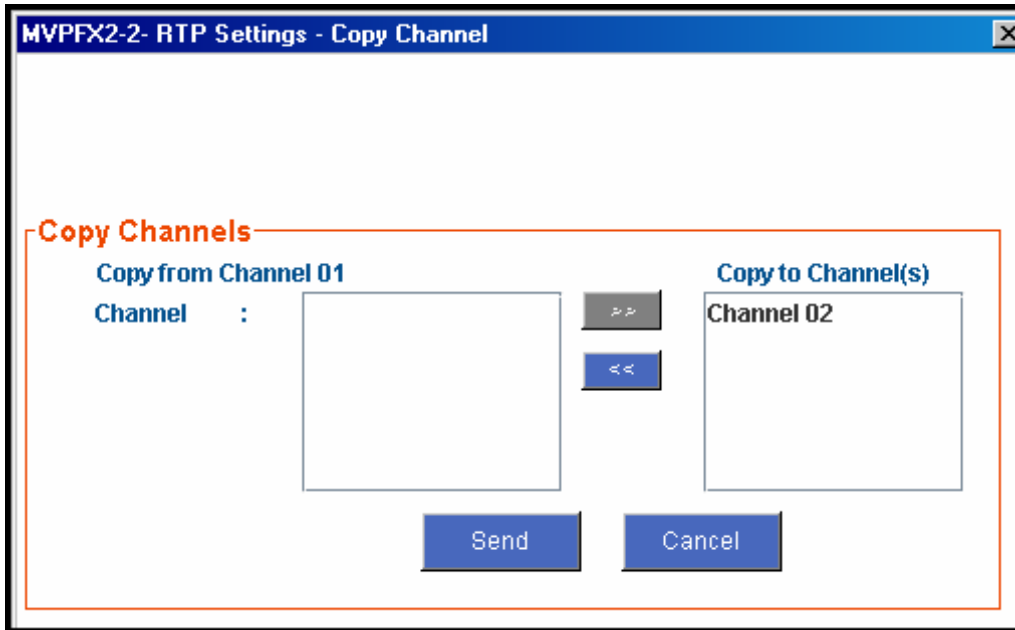
Default

Packetization rates can be set separately for each channel.

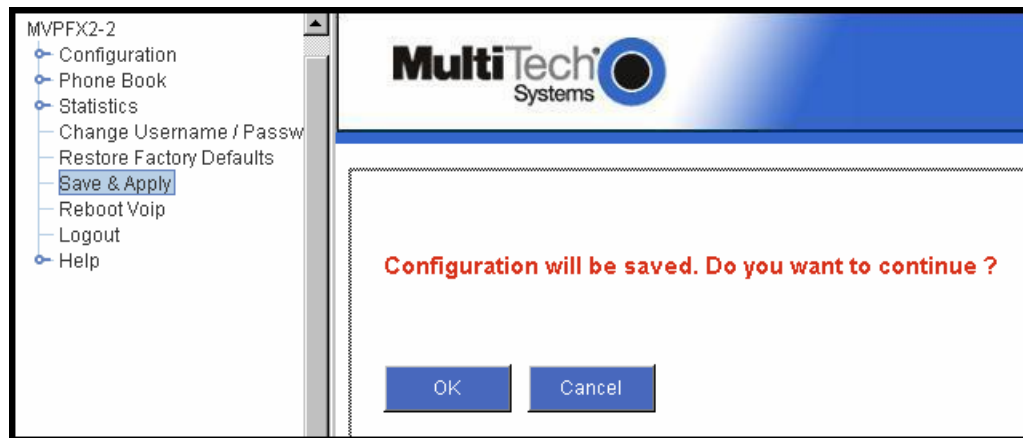
The table below presents the ranges and increments for packetization rates.

Packetization Ranges and Increments			
Coder Types	Range (in ms); {default value}		Increments (in ms)
G711 A-law	5-30	{ 30 }	5
G711 u-law	5-60	{ 60 }	10
G723	30-60	{ 60 }	10
G726	5-60	{ 60 }	10
G729	10-80	{ 80 }	20

Once the packetization rate has been set for one channel, it can be copied into other channels by using the Copy Channel screen.



13. **Saving the MultiVOIP Configuration.** When values have been set for all of the MultiVOIP's various operating parameters, click on **Save & Apply** in the sidebar.



**NOTE:** It is possible to return all parameters to their factory default values by using the Restore Factory Defaults command. By restoring factory-default values, this command will negate all configuration work that has been done.

---

## **Chapter 6: Phonebook Configuration**

## Configuring MultiVOIP Phonebooks

When a VoIP serves a PBX system, it's important that the operation of the VoIP be transparent to the telephone end user. That is, the VoIP should not entail the dialing of extra digits to reach users elsewhere on the network that the VoIP serves. On the contrary, VOIP service more commonly reduces dialed digits by allowing users (served by PBXs in facilities in distant cities) to dial their co-workers with 3-, 4-, or 5-digit extensions as if they were in the same facility.

Furthermore, the setup of the VoIP generally should allow users to make calls on a non-toll basis to any numbers accessible without toll by users at all other locations on the VoIP system. Consider, for example, a company with VOIP-equipped offices in New York, Miami, and Los Angeles, each served by its own PBX. When the VOIP phone books are set correctly, personnel in the Miami office should be able to make calls without toll not only to the company's offices in New York and Los Angeles, but also to any number that's local in those two cities.

To achieve transparency of the VoIP telephony system and to give full access to all types of non-toll calls made possible by the VOIP system, the VoIP administrator must properly configure the "Outbound" and "Inbound" phone-books of each VoIP in the system.

The "Outbound" phonebook for a particular VoIP unit describes the dialing sequences required for a call to originate locally (typically in a PBX in a particular facility) and reach any of its possible destinations at remote VoIP sites, including non-toll calls completed in the PSTN at the remote site.

The "Inbound" phonebook for a particular VoIP unit describes the dialing sequences required for a call to originate remotely from any other VOIP sites in the system, and to terminate on that particular VOIP.

Briefly stated, *the MultiVOIP's Outbound phonebook lists the phone stations it can call; its Inbound phonebook describes the dialing sequences that can be used to call that MultiVOIP and how those calls will be directed.* (Of course, the phone numbers are not literally "listed" individually, but are, instead, described by rule.)

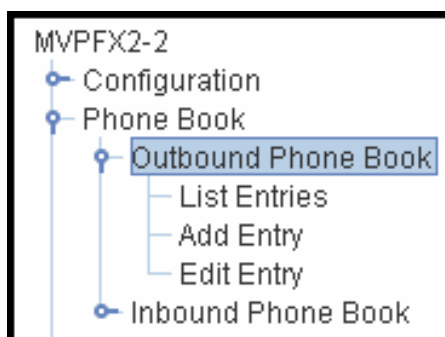
Consider two types of calls in the three-city system described above: (1) calls originating from the Miami office and terminating in the New York (Manhattan) office, and (2) calls originating from the Miami office and terminating in New York City but off the company's premises in an adjacent area code, an area code different than the company's office but still a local call from that office (e.g., Staten Island).

The first type of call requires an entry in the Outbound PhoneBook of the Miami VOIP and a coordinated entry in the Inbound phonebook of the New York VOIP. These entries would allow the Miami caller to dial the New York office as if its phones were extensions on the Miami PBX.

The second type of call similarly requires an entry in the Outbound PhoneBook of the Miami VOIP and a coordinated entry in the Inbound Phonebook of the New York VOIP. However, these entries will be longer and more complicated. Any Miami call to New York City local numbers will be sent through the VOIP system rather than through the regular toll public phone system (PSTN). But the phonebook entries can be arranged so that the VOIP system is transparent to the Miami user, such that even though that Miami user dials the New York City local number just as they would through the public phone system, that call will still be completed through the VOIP system.

This PhoneBook Configuration procedure is brief, but it is followed by an example case. For many people, the example case may be easier to grasp than the procedure steps. Configuration is not difficult, but all phone number sequences and other information must be entered exactly; otherwise connections will not be made.

Phonebook configuration screens are accessed using the sidebar menu.



1. Select **Outbound Phone Book/List Entries**.

Fields in the “Details” section describe various SIP parameters.

Current Permission: Read/Write

**Outbound Phone Book**

Destination Pattern	IP Address	Protocol	Description
Any Number	243 . 4 . 14 . 72	SIP	voip-remote1

Number of Entries:

**Details**

Remove Prefix	<input type="text"/>	Add
Add Prefix	<input type="text"/>	Edit
SIP Proxy Server	not used	Delete
SIP port	5060	Close
Transport Protocol	TCP	
SIP URL	<input type="text"/>	

Click **Add**.

2. The **Outbound Phone Book Add Entry** screen appears.

The screenshot shows a configuration window titled "Outbound Phone Book Add Entry" with a blue header bar indicating "Current Permission: Read/Write". The window contains several sections:

- Phone Number Details:** A section with a blue border containing:
  - Accept Any Number**
  - Destination Pattern:** Text input field containing "365"
  - Total Digits:** Text input field containing "0"
  - Remove Prefix:** Text input field
  - Add Prefix:** Text input field
- IP Address:** Text input field
- Description:** Text input field
- SIP:** A section with a blue border containing:
  - Use Proxy**
  - Transport Protocol:** Radio buttons for **TCP** (selected) and **UDP**
  - SIP Port Number:** Text input field containing "5060"
  - SIP URL:** Text input field

On the right side of the window, there are two blue buttons: **OK** and **Cancel**.

Enter Outbound PhoneBook data for your MultiVOIP unit.



The fields of the **Outbound Phone Book Add Entry** screen are described in the table below.

<b>Outbound Phone Book Add Entry screen: Field Definitions</b>		
<b>Field Name</b>	<b>Values</b>	<b>Description</b>
Accept Any Number	Y/N	<p>When checked, “Any Number” appears as the value in the Destination Pattern field.</p> <p>The <b>Any Number</b> feature works differently depending on whether or not an external SIP Proxy routing device is used.</p> <p><b>When no external routing device is used.</b> If <b>Any Number</b> is selected, calls to phone numbers not matching a listed Destination Pattern will be directed to the <b>IP Address</b> in the Outbound Phone Book Add Entry screen. “Any Number” can be used in addition to one or more Destination Patterns.</p> <p><b>When external routing device is used.</b> If <b>Any Number</b> is selected, calls to phone numbers not matching a listed Destination Pattern will be directed to the external SIP proxy routing device. The IP Address of the external routing device must be set in the Phone Book Configuration screen.</p>

<b>Outbound Phone Book Add Entry screen: Field Definitions (cont'd)</b>		
<b>Field Name</b>	<b>Values</b>	<b>Description</b>
Destination Pattern	prefixes, area codes, exchanges, line numbers, extensions	Defines the beginning of dialing sequences for calls that will be connected to another VOIP in the system. Numbers beginning with these sequences are diverted from the PTSN and carried on Internet or other IP network.
Total Digits	as needed	<i>This field currently disabled.</i> Number of digits the phone user must dial to reach specified destination.
Remove Prefix	dialed digits	Portion of dialed number to be removed before completing call to destination.
Add Prefix	dialed digits	Digits to be added before completing call to destination.
IP Address	n.n.n.n for n = 0-255	The IP address to which the call will be directed if it begins with the destination pattern given.
Description	alpha-numeric	Describes the facility or geographical location at which the call will be completed.

Outbound Phone Book Add Entry screen: Field Definitions (cont'd)		
Field Name	Values	Description
<b>SIP Fields</b>		
Use Proxy	Y/N	Select if proxy server is used.
Transport Protocol	TCP or UDP	Voip administrator must choose between UDP and TCP transmission protocols. UDP is a high-speed, low-overhead connectionless protocol where data is transmitted without acknowledgment, guaranteed delivery, or guaranteed packet sequence integrity. TCP is slower connection-oriented protocol with greater overhead, but having acknowledgment and guarantees delivery and packet sequence integrity.
SIP Port Number	5060 or other  *See RFC 3087 ("Control of Service Context using SIP Request-URI," by the Network Working Group).	The SIP Port Number is a UDP logical port number. The voip will "listen" for SIP messages at this logical port. If SIP is used, 5060 is the default, standard, or "well known" port number to be used. If 5060 is not used, then the port number used is that specified in the SIP Request URI (Universal Resource Identifier).
SIP URL	<i>sip.userphone</i> @ <i>hostserver</i> , where "userphone" is the telephone number and "hostserver" is the domain name or an address on the network	Looking similar to an email address, a <b>SIP URL</b> identifies a user's address. In SIP communications, each caller or callee is identified by a SIP url: sip:user_name@host_name. The format of a sip url is very similar to an email address, except that the "sip:" prefix is used.

3. Select **Inbound PhoneBook | List Entries.**

Current Permission: Read/Write

**Inbound Phone Book**

Remove Prefix	Add Prefix
181	
182	
234	6

Number of Entries

**Details**

Channel No

Description

**Registration Options**

Register With SIP Proxy - Disabled

Click **Add**.

4. The **Inbound Phone Book Add Entry** screen appears.

Current Permission: Read/Write

**Inbound Phone Book Add Entry**

Accept Any Number

Remove Prefix

Add Prefix

Channel Number **Hunting** ▼

Description

**Registration Options**

Register With SIP Proxy

OK

Cancel

Enter Inbound PhoneBook data for your MultiVOIP. The fields of the Inbound Phone Book Add Entry screen are described in the table below.

<b>Inbound Phone Book Add Entry screen: Field Definitions</b>		
<b>Field Name</b>	<b>Values</b>	<b>Description</b>
Accept Any Number	<b>Values:</b> Y/N	<b>Description:</b> When checked, "Any Number" appears as the value in the Remove Prefix field.  The <b>Any Number</b> feature of the Inbound Phone Book does not work when an external routing device is used (Proxy for SIP protocol).  <b>When no external routing device is used.</b> If <b>Any Number</b> is selected, calls received from phone numbers not matching a listed Prefix (shown in the Remove Prefix column of the Inbound Phone Book) will be admitted into the voip on the channel listed in the <b>Channel Number</b> field. "Any Number" can be used in addition to one or more Prefixes.
Remove Prefix	dialed digits	portion of dialed number to be removed before completing call to destination (often a local PBX)
Add Prefix	dialed digits	digits to be added before completing call to destination (often a local PBX)
Channel Number	1-24, or "Hunting" or 1-2 or 1-4 or 1-8 depending on model	Channel number to which the call will be assigned as it enters the local telephony equipment (often a local PBX). "Hunting" directs the call to any available channel.
Description	--	Describes the facility or geographical location at which the call originated.
Register with SIP Proxy	Y/N  *Must be enabled in Call Signaling Configuration screen.	When checked, the value in the <b>Remove Prefix</b> field will be registered with the external SIP proxy routing device. The IP address of the external routing device must be set in the Phone Book Configuration screen.  In a SIP voip system, gateways can register with the SIP Proxy.

5. When your Outbound and Inbound PhoneBook entries are completed, click on **Save and Apply** in the sidebar menu to save your configuration.

You can change your configuration at any time as needed for your system.

There are two “Edit Entry” screens for revising outbound and inbound phone book entries.

Current Permission: Read/Write

### Outbound Phone Book Edit Entry

#### Phone Number Details

Accept Any Number

Destination Pattern

Total Digits

Remove Prefix

Add Prefix

IP Address

Description

#### SIP

Use Proxy

**Transport Protocol**

TCP  UDP

SIP Port Number

SIP URL

OK

Cancel

Next Entry

Previous Entry

The “Next Entry” and “Previous Entry” buttons allow you to go from one entry to the next without interruption. You must click **OK** to confirm the change before moving on to the next entry.

Current Permission: Read/Write

### Inbound Phone Book Edit Entry

Accept Any Number

Remove Prefix: 401

Add Prefix:

Channel Number: Channel 01

Description:

OK

Cancel

Next Entry

Previous Entry

#### Registration Options

Register With SIP Proxy

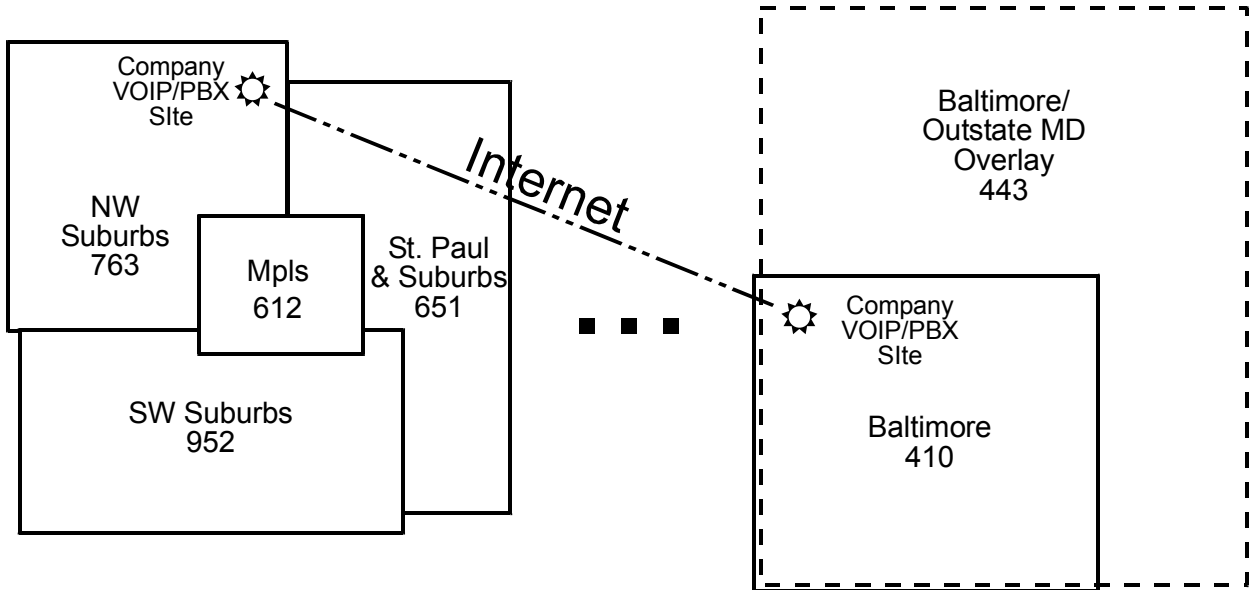


## Phonebook Examples

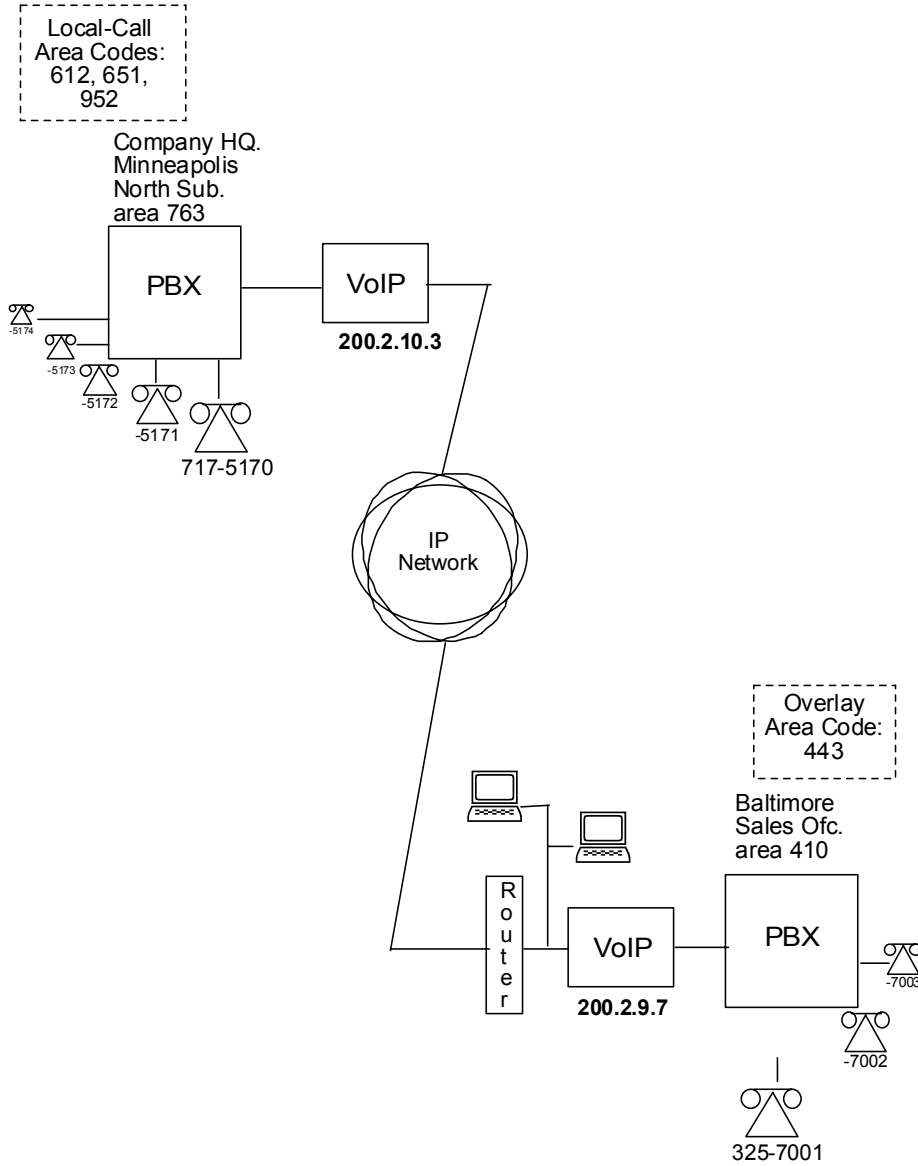
The following example demonstrates how Outbound and Inbound PhoneBook entries work in a situation of multiple area codes. Consider a company with offices in Minneapolis and Baltimore. The system depicted is SIP. However, the phonebook entries presented are still applicable for SIP systems.

### 2 Site Example

Notice first the area code situation in those two cities: Minneapolis's local calling area consists of multiple adjacent area codes; Baltimore's local calling area consists of a base area code plus an overlay area code.



An outline of the equipment setup in both offices is shown below.



The screen below shows Outbound PhoneBook entries for the VOIP located in the company’s Baltimore facility.

Current Permission: Read/Write

**Outbound Phone Book** — { Baltimore voip unit }

Destination Pattern	IP Address	Protocol	Description
1612	200.002.010.003	SIP	Minneapolis
1651	200.002.010.003	SIP	St. Paul
1763	200.002.010.003	SIP	Minneapolis, N. Suburbs
1952	200.002.010.003	SIP	Minneapolis, S. Suburbs

Number of Entries: 4

**Details**

Remove Prefix	<input type="text"/>	Add
Add Prefix	<input type="text"/>	Edit
SIP Proxy Server	not used	Delete
SIP port	5060	Close
Transport Protocol	TCP	
SIP URL	<input type="text"/>	

The entries in the Minneapolis VOIP’s Inbound PhoneBook match the Outbound PhoneBook entries of the Baltimore VOIP, as shown below.

Current Permission: Read/Write

**Inbound Phone Book** — { Minneapolis voip unit }

Remove Prefix	Add Prefix
1612	9,612
1651	9,651
1763	9,
17637175	5
1952	9,952

Number of Entries: 5

**Details**

Channel No	1	Add
Description	<input type="text"/>	Edit
<b>Registration Options</b>		Delete
Register With SIP Proxy - Disabled		Close

To call the Minneapolis/St. Paul area, a Baltimore employee must dial eleven digits. (In this case, we are assuming that the Baltimore PBX does not require an "8" or "9" to seize an outside phone line.)

If a Baltimore employee dials any phone number in the 612 area code, the call will automatically be handled by the company's voip system. Upon receiving such a call, the Minneapolis voip will remove the digits "1612". But before the suburban-Minneapolis voip can complete the call to the PSTN of the Minneapolis local calling area, it must dial "9" (to get an outside line from the PBX) and then a comma (which denotes a pause to get a PSTN dial tone) and then the 10-digit phone number which includes the area code (612 for the city of Minneapolis; which is different than the area code of the suburb where the PBX is actually located -- 763).

A similar sequence of events occurs when the Baltimore employee calls number in the 651 and 952 area codes because number in both of these area codes are local calls in the Minneapolis/St. Paul area.

The simplest case is a call from Baltimore to a phone within the Minneapolis/St. Paul area code where the company's voip and PBX are located, namely 763. In that case, that local voip removes 1763 and dials 9 to direct the call to its local 7-digit PSTN.

Finally, consider the longest entry in the Minneapolis Inbound Phonebook, "17637175. Note that the main phone number of the Minneapolis PBX is 763-717-5170. The destination pattern 17637175 means that all calls to Minneapolis employees will stay within the suburban Minneapolis PBX and will not reach or be carried on the local PSTN.

Similarly, the Inbound PhoneBook for the Baltimore VOIP (shown first below) generally matches the Outbound PhoneBook of the Minneapolis VOIP (shown second below).

Current Permission: Read/Write

**Inbound Phone Book** — { Baltimore voip unit }

Remove Prefix	Add Prefix	
1410	9	▲
14103257	7	☰
1443	9,443	▼

Number of Entries

**Details**

Channel No

Description

**Registration Options**

Register With SIP Proxy - Disabled

Notice the extended prefix to be removed: 14103257. This entry allows Minneapolis users to contact Baltimore co-workers as though they were in the Minneapolis facility, using numbers in the range 7000 to 7999.

Note also that a comma (as in the entry 9,443) denotes a delay in dialing. A one-second delay is commonly used to allow a second dial tone to be generated for calls going outside of the facility's PBX system.

The Outbound PhoneBook for the Minneapolis VOIP is shown below. The third destination pattern, “7” facilitates reception of co-worker calls using local-appearing-extensions only. In this case, the “Add Prefix” field value for this phonebook entry would be “1410325” .

Current Permission: Read/Write

**Outbound Phone Book** — { *Minneapolis voip unit* }

Destination Pattern	IP Address	Protocol	Description
1410	200.002.009.007	SIP	Baltimore
1443	200.002.009.007	SIP	Baltimore overlay
7	200.002.009.007	SIP	Baltimore Office Extensions

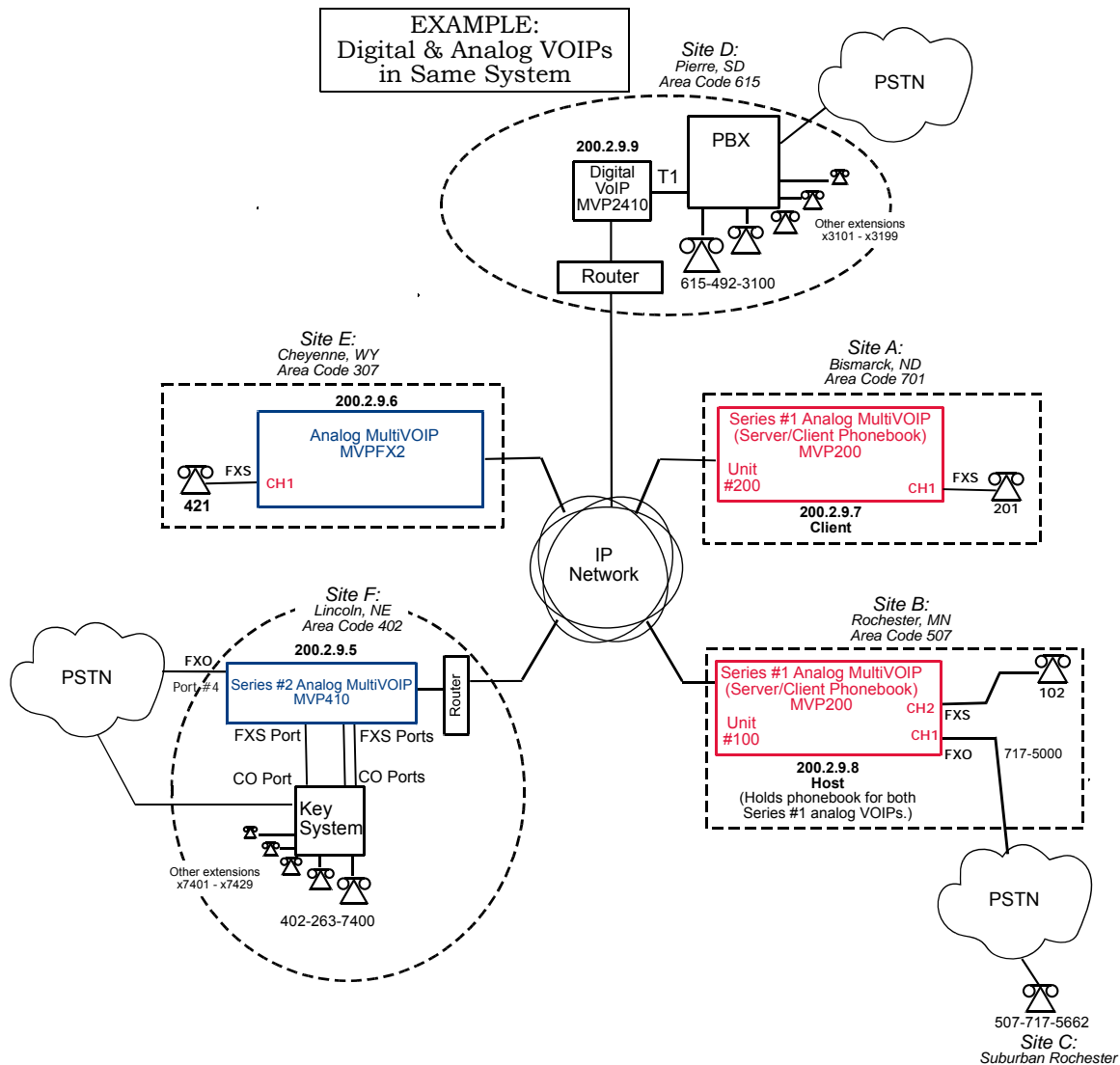
Number of Entries:

**Details**

Remove Prefix	<input type="text"/>	Add
Add Prefix	<input type="text"/>	Edit
SIP Proxy Server	not used	Delete
SIP port	5060	Close
Transport Protocol	TCP	
SIP URL	<input type="text"/>	

## Configuring Mixed Digital/Analog VOIP Systems

Analog MultiVOIP units, like the MVPFX2-2/4/8 units and the MVP-210/410/810/410SS/810SS units are compatible with digital MultiVOIP units like the MVP2410. In many cases, digital and analog VOIP units will appear in the same telephony/IP system. In addition to MVP-210/410/810 MultiVOIP units (Series II units), legacy analog VOIP units (Series I units made by MultiTech) may be included in the system, as well. When legacy VOIP units are included, the VOIP administrator must handle two styles of phonebooks in the same VOIP network. The diagram below shows a small-scale system of this kind: one digital VOIP (the MVP2410) operates with a Series II analog VOIP (an MVP410), and two Series I legacy VOIPs (two MVP200 units) and an MVPFX2-2.



The Series I analog VOIP phone book resides in the “Host” VOIP unit at Site B. It applies to both of the Series I analog VOIP units.

The Series II analog MultiVOIP (the MVP410) requires its own inbound and outbound phonebooks. The MVP2410 digital MultiVOIP requires its own inbound and outbound phonebooks, also, as does the MVPFX2-2 unit.

These seven phone books are shown below.

<b>Phone Book for Series I Analog VOIP Host Unit (Site B)</b>			
VOIP Dir # -OR- Destination Pattern	IP Address	Channel	Comments
102	200.2.9.8	2	Site B, FXS channel.
101	200.2.9.8	1	Site B, FXO channel.
421	200.2.9.6	1	Site E FXS channel.
201	200.2.9.7	1	Site A, FXS channel.
1615 xxx xxxx	200.2.9.9	0 (Note 2.)	Gives remote voip users access to local PSTN of Site D (Pierre, SD, area code 615).
3xxx  (Note 1.)	200.2.9.9	0	Allows remote voip users to call all PBX extensions at Site D (Pierre, SD) using only four digits.
1402	200.2.9.5	0	Gives remote voip users access to local PSTN of Site F (Lincoln, NE; area code 402).
140226374 (Note 1) (Note 3)	200.2.9.5	0	Gives remote voip users access to key phone system extensions at Site F (Lincoln).
<p>Note 1. The "x" is a wildcard character.</p> <p>Note 2. By specifying "Channel 0," we instruct the MVP2400/2410 to choose any available data channel to carry the call.</p> <p>Note 3. Note that Site F key system has only 30 extensions (x7400-7429). This destination pattern (140226374) actually directs calls to 402-263-7430 through 402-263-7499 into the key system, as well. This means that such calls, which belong on the PSTN, cannot be completed. In some cases, this might be inconsequential because an entire exchange (fully used or not) might have been reserved for the company or it might be unnecessary to reach those numbers. However, to specify only the 30 lines actually used by the key system, the destination pattern 140226374 would have to be replaced by three other destination patterns, namely 1402263740, 1402263741, and 1402263742. In this way, calls to 402-263-7430 through 402-263-7499 would be properly directed to the PSTN. In the Site D outbound phonebook, the 30 lines are defined exactly, that is, without making any adjacent phone numbers unreachable through the voip system.</p>			



Outbound Phone Book for MVP2410 Digital VOIP (Site D)				
Destin. Pattern	Remove Prefix	Add Prefix	IP Address	Comment
201			200.2.9.7	To originate calls to Site A (Bismarck).
1507	1507	101# Note 3.	200.2.9.8	To originate calls to Rochester local PSTN using the FXO channel (channel #1) of the Site B VOIP.
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP.
421			200.2.9.6	Calls to Site E (Cheyenne).
1402			200.2.9.5	Calls to Lincoln area local PSTN (via FXO channel, CH4, of the Site F VOIP).
1402 263 740			200.2.9.5	Calls to extensions (thirty) of key system at Site F (Lincoln). Human operator or auto-attendant is needed to complete these calls.
1402 263 741		200.2.9.5		
1402 263 742		200.2.9.5		
Note 3. The pound sign (“#”) is a delimiter separating the VOIP number from the standard telephony phone number.				

Inbound Phonebook for MVP2410 Digital VOIP (Site D)			
Remove Prefix	Add Prefix	Channel Number	Comment
1615	9, Note 4. Note 5.	0 (hunting)	Allows phone users at remote voip sites to call non-toll numbers within the Site D area code (615; Pierre, SD) over the VOIP network.
1615 49231	31	0, (hunting)	Allows voip calls directly to employees at Site D (at extensions x3101 to x3199).
Note 4. “9” gives PBX station users access to outside line.			
Note 5. The comma represents a one-second pause, the time required for the user to receive a dial tone on the outside line (PSTN). The comma is only allowed in the Inbound phonebook.			

<b>Outbound Phone Book for MVP410 Analog VOIP (Site F)</b>				
<b>Destin. Pattern</b>	<b>Remove Prefix</b>	<b>Add Prefix</b>	<b>IP Address</b>	<b>Comment</b>
201			200.2.9.7	To originate calls to Site A (Bismarck).
1507	1507	101# Note 3.	200.2.9.8	To originate calls to any PSTN phone in Rochester area using the FXO channel (channel #1) of the Site B VOIP.
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP (Rochester).
421			200.2.9.6	Calls to Site E (Cheyenne).
1615			200.2.9.9	Calls to Pierre area PSTN via Site D PBX.
31		1615 492	200.2.9.9	Calls to Pierre PBX extensions with four digits.
Note 3. The pound sign (“#”) is a delimiter separating the VOIP number from the standard telephony phone number.				

<b>Inbound Phonebook for MVP410 Analog VOIP (Site F)</b>			
<b>Remove Prefix</b>	<b>Add Prefix</b>	<b>Channel Number</b>	<b>Comment</b>
1402		4	Access to Lincoln local PSTN by users at remote VOIP locations via FXO port at Site F.
1402 263740	740	0 (hunting)	Gives remote voip users access to extension of key phone system at Site F (Lincoln). Because call is completed at key system, abbreviated dialing (4 digits) is not workable. Human operator or auto-attendant is needed to complete these calls.
1402 263741	741	0 (hunting)	
1402 263742	742	0 (hunting)	

<b>Outbound Phone Book for MVPFX2 Analog VOIP (Site E)</b>				
<b>Destin. Pattern</b>	<b>Remove Prefix</b>	<b>Add Prefix</b>	<b>IP Address</b>	<b>Comment</b>
201			200.2.9.7	To originate calls to Site A.
1507	1507	101# Note 3.	200.2.9.8	To originate calls to any PSTN phone in Rochester area using the FXO channel (channel #1) of the Site B VOIP.
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP.
1402			200.2.9.5	Calls to Lincoln area PSTN (via FXO channel, CH4, of the Site F VOIP).
7		1402 263	200.2.9.5	Calls to Lincoln key extensions with four digits.
1615			200.2.9.9	Calls to Pierre area PSTN via Site D PBX.
31		1615 492	200.2.9.9	Calls to Pierre PBX extensions with four digits.
Note 3. The pound sign (“#”) is a delimiter separating the VOIP number from the standard telephony phone number.				

<b>Inbound Phonebook for MVPFX2 Analog VOIP (Site E)</b>			
<b>Remove Prefix</b>	<b>Add Prefix</b>	<b>Channel Number</b>	<b>Comment</b>
421		1	

## Call Completion Summaries

### Site A calling Site C, Method 1

1. Dial 101.
2. Hear dial tone from Site B.
3. Dial 7175662.
4. Await completion. Talk.

### Site A calling Site C, Method 2

1. Dial 101#7175662
2. Await completion. Talk.

---

Note: Series I analog VOIP gateways will allow completion by Method 2.  
Others will not.

---

### Site C calling Site A

1. Dial 7175000.
2. Hear dial tone from Site B VOIP.
3. Dial 201.
4. Await completion. Talk.

**Site D calling Site C**

1. Dial 915077175662.
2. "9" gets outside line. On some PBXs, an "8" may be used to direct calls to the VOIP, while "9" directs calls to the PSTN. However, some PBX units can be programmed to identify the destination patterns of all calls to be directed to the VOIP.
3. PBX at Site D is programmed to divert all calls made to the 507 area code and exchange 717 into the VOIP network. (It would also be possible to divert all calls to all phones in area code 507 into the VOIP network, but it may not be desirable to do so.)
4. The MVP2410 removes the prefix "1507" and adds the prefix "101#" for compatibility with the analog MultiVOIP's phonebook scheme. The "#" is a delimiter separating the analog VOIP's phone number from the digits that the analog VOIP must dial onto its local PSTN to complete the call. The digits "101#7175662" are forwarded to the Site B analog VOIP.
5. The call passes through the IP network (in this case, the Internet).
6. The call arrives at the Site B VOIP. This analog VOIP receives this dialing string from the MVP2410: 101#7175662. The analog VOIP, seeing the "101" prefix, uses its own channel #1 (an FXO port) to connect the call to the PSTN. Then the analog VOIP dials its local phone number 7175662 to complete the call.

**Site D calling Site F**

A voip call from Pierre PBX to extension 7424 on the key telephone system in Lincoln, Nebraska.

A. The required entry in the Pierre Outbound Phonebook to facilitate origination of the call, would be 1402263742. The call would be directed to the Lincoln voip's IP address, 200.2.9.5.

(Generally on such a call, the caller would have to dial an initial "9." But typically the PBX would not pass the initial "9" to the voip. If the PBX *did* pass along that "9" however, its removal would have to be specified in the local Outbound Phonebook.)

B. The corresponding entry in the Lincoln Inbound Phonebook to facilitate completion of the call would be

1402263742 for calls within the office at Lincoln

1402 for calls to the Lincoln local calling area (PSTN).

**Call Event Sequence**

1. Caller at Pierre dials 914022637424.
2. Pierre PBX removes "9" and passes 14022637424 to voip.
3. Pierre voip passes remaining string, 14022637424 on to the Lincoln voip at IP address 200.2.9.5.
4. The dialed string matches an inbound phonebook entry at the Lincoln voip, namely 1402263742.
5. The Lincoln voip rings one of the three FXS ports connected to the Lincoln key phone system.
6. The call will be routed to extension 7424 either by a human receptionist/operator or to an auto-attendant (which allows the caller to specify the extension to which they wish to be connected).

## Site F calling Site D

A voip call from a Lincoln key extension to extension 3117 on the PBX in Pierre, South Dakota.

A. The required entry in the Lincoln Outbound Phonebook to facilitate origination of the call, would be “31”. The string “1615492” would have to be added as a prefix. The call would be directed to the Pierre voip’s IP address, 200.2.9.9.

B. The corresponding entry in the Pierre Inbound Phonebook to facilitate completion of the call would be 161549231.

1. Caller at Lincoln picks up phone receiver, presses button on key phone set. This button has been assigned to a particular voip channel (any one of the three FXS ports).
2. The caller at Lincoln hears dial tone from the Lincoln voip.
3. The caller at Lincoln dials 3117.
4. The Lincoln voip adds the prefix 1615492 and sends the entire dialing string, 16154923117, to the Pierre voip at IP address 200.2.9.9.
5. The Pierre voip matches the called digits 16154923117 to its Inbound Phonebook entry “161549231” .
6. The Pierre PBX dials extension 3117 in the office at Pierre.

## Variations in PBX Characteristics

The exact dialing strings needed in the Outbound and Inbound Phonebooks of the MultiVOIP units will depend on the capabilities of the PBX. Some PBXs require trunk access codes (like an “8” or “9” to access an outside line or to access the VOIP network). Other PBXs can automatically distinguish between intra-PBX calls, PSTN calls, and VOIP calls.

Some PBX units can also insert digits automatically when they receive certain dialing strings from a phone station. For example, a PBX may be programmable to insert automatically the three-digit VOIP identifier strings into calls to be directed to analog VOIPs.

The MultiVOIP offers complete flexibility for inter-operation with PBX units so that a coherent dialing scheme can be established to connect a company’s multiple sites together in a way that is convenient and intuitive for phone users. When working together with modern PBX units, the presence of the MultiVOIP can be completely transparent to phone users within the company.

---

## **Chapter 7: Operation and Maintenance**



## Operation and Maintenance Summary

There are several groups of software screens that facilitate basic operation and maintenance of the MultiVOIP. The following commands and functions are accessible on the sidebar menu and they allow you to:

- (a) view a summary of important **System Information** (like software version levels),
- (b) track **Call Progress Statistics**,
- (c) track **IP Statistics**,
- (d) Change the MultiVOIP's **Username and Password**,
- (e) **Restore Factory Default** values for most operating parameters (except the voip's IP address and its phonebook entries),
- (f) **Save & Apply** new settings,
- (g) **Reboot** the MultiVOIP, and
- (h) **Log out** of the MultiVOIP.

Another group of operation & maintenance functions concerns the upgrading of certain important system files that reside on the MultiVOIP. This upgrading can be done either by an FTP transfer through a browser or by a TFTP transfer done through the MultiVOIP's Console port.

This chapter ends with a discussion of how a SysLog application program can facilitate logging of traffic on the MultiVOIP network.

## System Information screen

This screen presents vital system information at a glance. Its primary use is in troubleshooting. This screen is accessible by clicking **System Information** in the sidebar menu.

The screenshot displays the 'System Information' screen. At the top, a blue header bar shows 'Current Permission: Read/Write'. Below this, the title 'System Information' is written in orange. A blue-bordered box contains the 'Version Information' section, which lists several system components and their versions. Below this box, the 'MAC Address' and 'Uptime' are displayed. At the bottom center, there is a blue 'Exit' button.

Version Information	
Boot Version	0.2.08
Firmware Version	13.01.11
Configuration Version	12.01.01
Phone Book Version	12.01.01
MSP Version	5.03
Kernel Version	V2_17_1.0

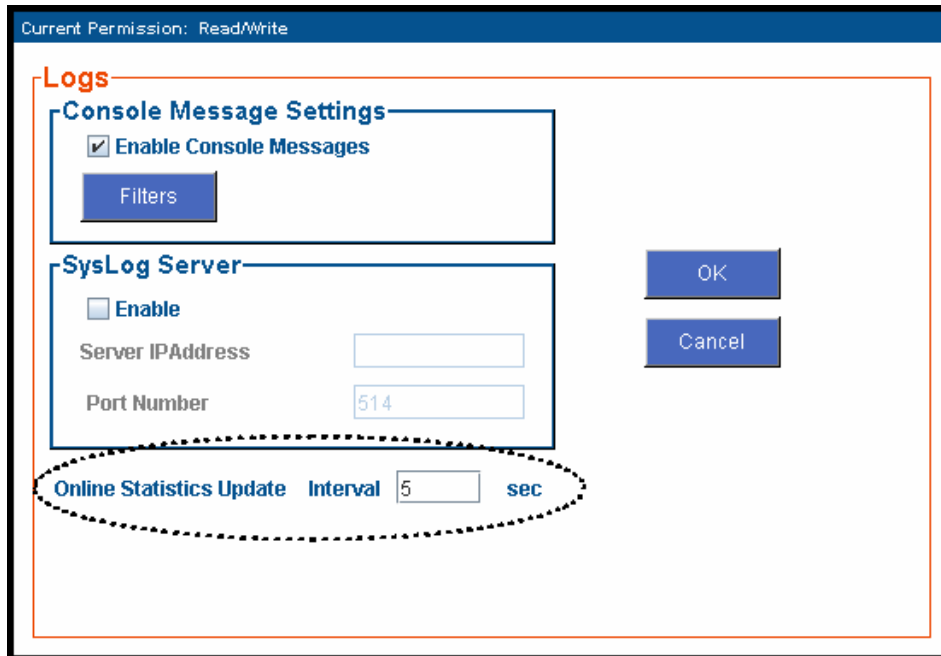
  

MAC Address	00:B6:C6:D6:E6:EE
Uptime	01:02:23:51

Exit

System Information Parameter Definitions		
Field Name	Values	Description
Boot Version	nn.nn alpha-numeric	Indicates the version of the code that is used at the startup (booting) of the voip. The boot code version is independent of the software version.
Firmware Version	nn.nn.nn alpha-numeric	Indicates the version of the MultiVOIP firmware.
Configuration Version	nn.nn. nn.nn alpha-numeric	Indicates the version of the MultiVOIP configuration software.
Phone Book Version	nn.nn alpha-numeric	Indicates the version of the MultiVOIP phone book being used.
MSP Version	nn.nn alpha-numeric	Version of DSP (digital signal processor) software used in MultiVOIP.
Kernel Version	Vn_nn_ n.n	Linux kernel version used in MultiVOIP.
Mac Address	numeric	Denotes the number assigned as the voip unit's unique Ethernet address.
Up Time	days: hours: mm:ss	Indicates how long the voip has been running since its last booting.

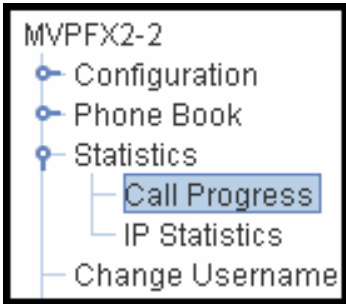

The frequency with which several administrative screens are updated (the System Information, Call Progress, and IP Statistics screens) is determined by a setting in the Logs/Traces screen.



## Statistics Screens

Ongoing operation of the MultiVOIP, whether it is in a MultiVOIP/PBX setting or MultiVOIP/telco-office setting, can be monitored for performance using the Statistics functions of the MultiVOIP software.

## About Call Progress

<b>Accessing Call-Progress Statistics</b>	
<b>Channel Icons (Main Screen Lower Left)</b> To access the Call Progress Details screen, click on “Call Progress” in the sidebar menu.	
	
Channel icons are green when data traffic is present, red when idle.	
Call progress details can be viewed by clicking on an icon (one for each channel) shown at the bottom of the web-browser screen.	

The Call Progress Details Screen

The screenshot shows a web interface titled "Call Progress Details" with a "Current Permission: Read/Write" header. It features a dropdown menu for "Channel" set to "Channel 01". Below this are two main sections: "Call Details" and "From - To Details".

**Call Details:**

- Duration: [input field]
- Mode: [input field]
- Voice coder: [input field]
- Packets sent: [input field]
- Packets Received: [input field]
- Bytes sent: [input field]
- Bytes Received: [input field]
- Packets Lost: [input field]
- Outbound Digits: [input field]
- Prefix Matched: [input field]

**From - To Details:**

- From: [input field] To: [input field]
- Gateway Name: [input field]
- IP Address: [input field]
- Options: [input field]

At the bottom, there are status indicators: "SC - Silence Compression" and "FEC - Forward Error Correction". The "Call Status" is "On Hook" with "Disconnect" and "Exit" buttons. The "Call Control Status" is [input field].

Call Progress Details: Field Definitions		
Field Name	Values	Description
Channel	1-n	Number of data channel or time slot on which the call is carried. This is the channel for which call-progress details are being viewed.
<b>Call Details</b>		
Duration	Hours: Minutes: Seconds	The length of the call in hours, minutes, and seconds (hh:mm:ss).
Mode	Voice or FAX	Indicates whether the call being described was a voice call or a FAX call.
Voice Coder	G.723, G.729, G.711, etc.	The voice coder being used on this call.

<b>Call Progress Details: Field Definitions</b>		
<b>Field Name</b>	<b>Values</b>	<b>Description</b>
<b>Packet Details</b>		
Packets Sent	integer value	The number of data packets sent over the IP network in the course of this call.
Packets Rcvd	integer value	The number of data packets received over the IP network in the course of this call.
Bytes Sent	integer value	The number of bytes of data sent over the IP network in the course of this call.
Bytes Rcvd	integer value	The number of bytes of data received over the IP network in the course of this call.
Packets Lost	integer value	The number of voice packets from this call that were lost while traversing the IP network.
Outbound Digits Sent	0-9, #, *	The digits transmitted by the MultiVOIP to the PBX/telco for this call.
Prefix Matched	specified dialing digits	Displays the dialed digits that were matched to a phonebook entry.

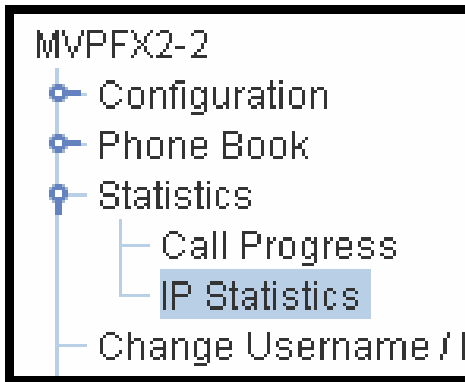
<b>Call Progress Details: Field Definitions (cont'd)</b>		
<b>From – To Details</b>		<b>Description</b>
<i>From</i> field	alphanumeric string	Description of calling party.
Gateway Name (from)	alphanumeric string	Identifier for the VOIP gateway that handled the origination of this call.
IP Address (from)	x.x.x.x, where x has a range of 0 to 255	IP address from which the call was received.
Options	SC	Displays VOIP transmission options in use on the current call. These may include Forward Error Correction or Silence Compression.
<i>To</i> field	alphanumeric string	Description of called party.
Gateway Name (to)	alphanumeric string	Identifier for the VOIP gateway that handled the completion of this call.
IP Address (to)	x.x.x.x, where x has a range of 0 to 255	IP address to which the call was sent.



Call Progress Details: Field Definitions (cont'd)		
Field Name	Values	Description
<b>Call Status</b> fields		
Silence Compression	SC	"SC" stands for Silence Compression. With <b>Silence Compression</b> enabled, the MultiVOIP will not transmit voice packets but instead will transmit SID (Silent Indication) packets when silence is detected. This feature reduces the amount of network bandwidth that is being used by the voice channel.
Call Status	on-hook, active	Shows condition of current call.
Disconnect	(command button)	Disconnects the selected call in progress.
Exit	(command button)	Use to exit Call Progress screen.

## About IP Statistics

To access the IP Statistics screen, click on "IP Statistics" in the sidebar menu.



### IP Statistics Screen

Current Permission: Read/Write

**IP Statistics**

IP Address:

<b>Total Packets</b>	
Transmitted: <input type="text" value="202050"/>	Received: <input type="text" value="202011"/>

<b>UDP Packets</b>	
Transmitted: <input type="text" value="55971"/>	Received: <input type="text" value="55927"/>
	Received with Errors: <input type="text" value="0"/>

<b>TCP Packets</b>	
Transmitted: <input type="text" value="146078"/>	Received: <input type="text" value="146084"/>
Retransmitted: <input type="text" value="1"/>	Received with Errors: <input type="text" value="0"/>

<b>RTP Packets</b>	
Transmitted: <input type="text" value="0"/>	Received: <input type="text" value="0"/>
	Received with Errors: <input type="text" value="0"/>

<b>RTCP Packets</b>	
Transmitted: <input type="text" value="0"/>	Received: <input type="text" value="0"/>
	Received with Errors: <input type="text" value="0"/>

IP Statistics: Field Definitions		
Field Name	Values	Description
		<b>UDP versus TCP.</b> (User Datagram Protocol versus Transmission Control Protocol). UDP provides unguaranteed, connectionless transmission of data across an IP network. By contrast, TCP provides reliable, connection-oriented transmission of data.

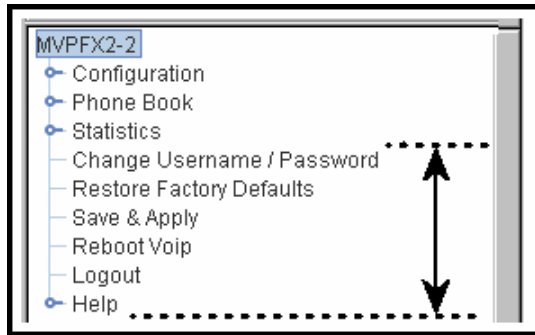
IP Statistics: Field Definitions		
Field Name	Values	Description
		<b>UDP versus TCP</b> (continued). Both TCP and UDP split data into packets called "datagrams." However, TCP includes extra headers in the datagram to enable retransmission of lost packets and reassembly of packets into their correct order if they arrive out of order. UDP does not provide this. Lost UDP packets are unretrievable; that is, out-of-order UDP packets cannot be reconstituted in their proper order.. Despite these obvious disadvantages, UDP packets can be transmitted much faster than TCP packets -- as much as three times faster. In certain applications, like audio and video data transmission, the need for high speed outweighs the need for verified data integrity. Sound or pictures often remain intelligible despite a certain amount of lost or disordered data packets (which appear as static).
IP Address	n.n.n.n 0 - 255	IP address of the MultiVOIP. If DHCP is enabled, the address assigned by the DHCP server will be displayed.
"Clear" button	--	Clears packet tallies from memory.
<b>Total Packets</b>		Sum of data packets of all types.
Transmitted	integer value	Total number of packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Total number of packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.

<b>IP Statistics: Field Definitions (cont'd)</b>		
<b>Field Name</b>	<b>Values</b>	<b>Description</b>
<b>Total Packets (cont'd)</b>		Sum of data packets of all types.
Received with Errors	integer value	Total number of error-laden packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
<b>UDP Packets</b>		User Datagram Protocol packets.
Transmitted	integer value	Number of UDP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of UDP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden UDP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
<b>TCP Packets</b>		Transmission Control Protocol packets.
Transmitted	integer value	Number of TCP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of TCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden TCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.

<b>IP Statistics: Field Definitions (cont'd)</b>		
<b>RTP Packets</b>		Voice signals are transmitted in Realtime Transport Protocol packets. RTP packets are a type or subset of UDP packets.
Transmitted	integer value	Number of RTP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of RTP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden RTP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
<b>RTCP Packets</b>		Realtime Transport Control Protocol packets convey control information to assist in the transmission of RTP (voice) packets. RTCP packets are a type or subset of UDP packets.
Transmitted	integer value	Number of RTCP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of RTCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden RTCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.

## General Operation Functions

Several basic software functions are accessible from the MultiVoip sidebar menu, as shown below.



Clicking on the option will bring up the corresponding screen.  
(Note that online Help has not yet been implemented.)

### Change Username/Password

To access the MultiVOIP web GUI, you must set up a username and a password.

#### Establishing a Username and Password

1. Go to the Change Username/Password screen.

Originally, all four fields on the screen will be blank.

2. Enter the desired values into the fields as follows:

Field Name	Your Entry	Comment
User Name:	_____	5 to 10 alphanumeric characters, case sensitive
Old Password:	_____	This will be blank at first. However, this field will be used if/when you revise the password.
New Password:	_____	5 to 13 characters, case sensitive
Reconfirm Password:	_____	Re-type new password exactly, letter for letter.

3. Click **OK**.

## About Passwords & Login/Logout from Specific Computers

The first time you access the MultiVOIP web GUI from a particular computer, you must enter the Username and Password.

If you close the browser without logging out (by clicking on Logout and clicking Yes), you will be able to access the MultiVOIP again *for the next 15 minutes* from that computer without logging in (that is, without entering your Username and Password). *Be aware of the security risk of closing the browser without logging out of the MultiVOIP program. If other users have access to that computer and if you want to prevent them from accessing the MultiVOIP unit, you must log out after using the MultiVOIP program.*

After you log out of the MultiVOIP web GUI, you will be required to enter your Username and Password to gain access to the MultiVOIP web GUI program.

Only one password can be assigned and it works for all MultiVOIP software functions (web browser GUI and FTP server – only the FTP Server function requires a User Name in addition to the password). After a password has been set, that password is required to access the MultiVOIP web browser GUI.

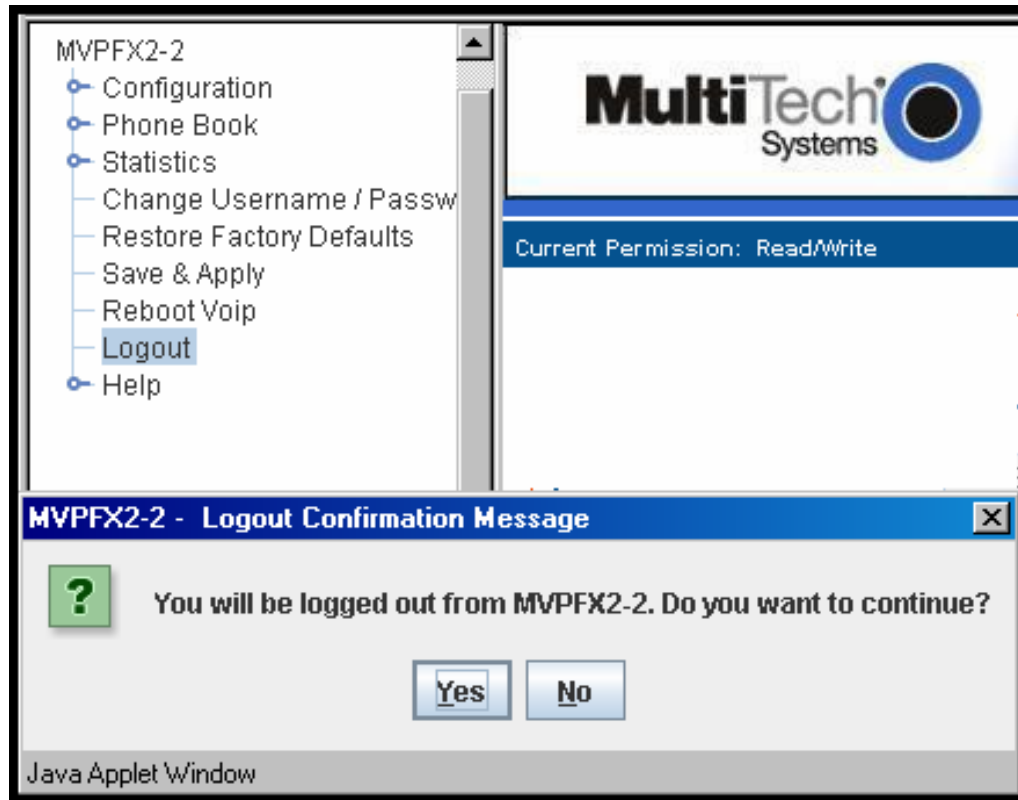
---

**NOTE:** Record your user name and password in a safe place. If the password is lost, forgotten, or unretrievable, the user must contact MultiTech Tech Support in order to resume use of the MultiVOIP web browser GUI.

---

## Logout

To log out, click on **Logout** in the sidebar menu and then click **Yes** to confirm your intention to log out of the MultiVOIP program.

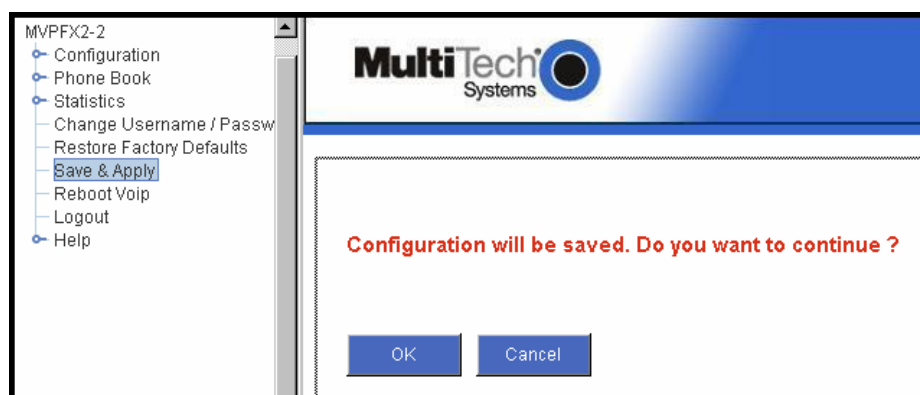


After you log out, you must enter your username and password again in order to access the MultiVOIP web GUI.

## Save & Apply

After you have changed MultiVOIP parameter values, you must invoke the **Save & Apply** command to make the changes permanent. When some parameters are changed, the MultiVOIP will reboot itself automatically; for other changes the automatic rebooting is not necessary.

To invoke the **Save & Apply** command, click on Save & Apply in the sidebar menu. Then click **OK** to confirm that you want to save the configuration.





## Reboot Voip

The Reboot Voip command allows you to reboot the MultiVOIP unit on demand. In the general course of operation, this command will not be needed. The command is included, however, to provide a remedy for situations when, by unexpected circumstances, code becomes corrupt and normal operation goes awry.

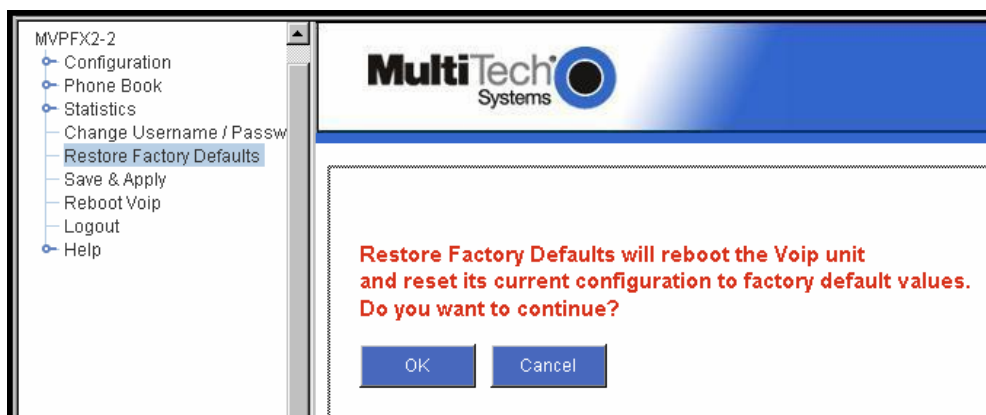
To invoke the Reboot Voip command, click on **Reboot Voip** in the sidebar menu and click **OK** to confirm your choice to reboot.



## Restore Factory Defaults

This command sets many MultiVOIP parameters back to their original values, as set in the factory. In the process of restoring factory default values, this command reboots the MultiVOIP unit. *The Restore Factory Defaults command does not alter user-specified IP Parameters settings; nor does it erase entries in the Inbound Phone Book or the Outbound Phone Book.*

To invoke the Restore Factory Defaults command, click on **Restore Factory Defaults** in the sidebar menu.



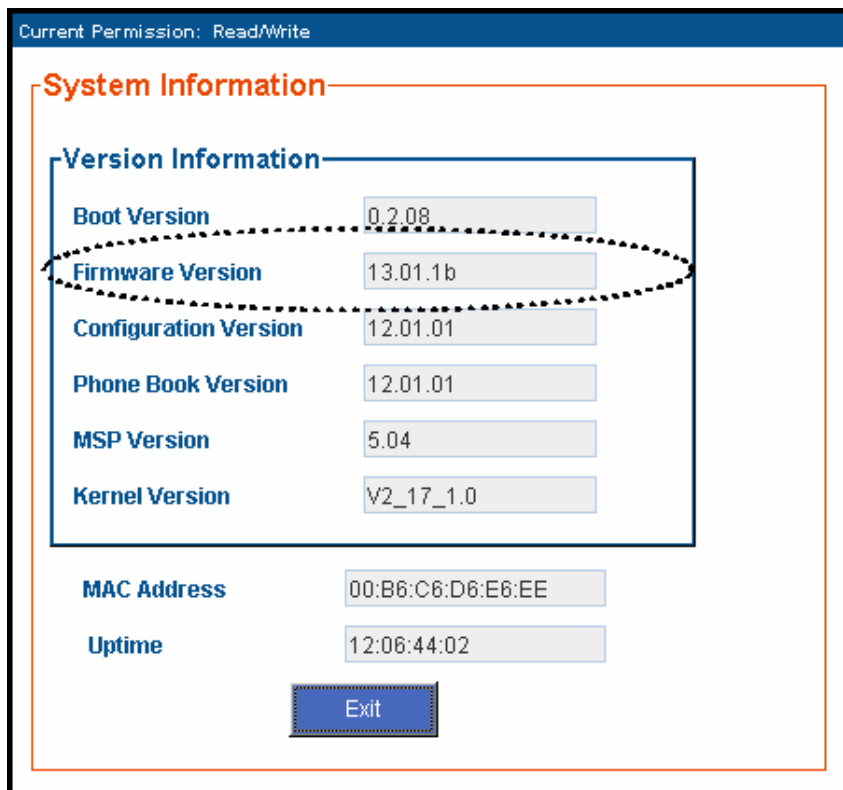
# Upgrading MultiVOIP Firmware

## Introduction

From time to time, a new version of the MVPFX2 firmware may be issued. When a new firmware version is issued, you can overwrite the MultiVOIP with the new code either by using a TFTP server or by using an FTP client and contacting the FTP server that resides on the MVPFX2 unit itself. It is also possible to update the firmware by a serial connection without FTP or TFTP, but that method of transfer is extremely slow. In the sections that follow, we present updating procedures using FTP (preferred) or TFTP.

## Identifying Current Firmware Version

Use the System Information screen to identify the current version of firmware on the MultiVOIP unit.



Obviously, if you are considering upgrading the firmware, you want to be sure that the new firmware is indeed at a higher revision level than the firmware currently on your MultiVOIP.

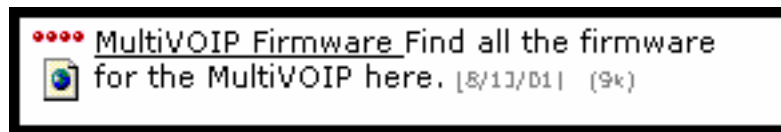
## Obtaining Updated Firmware

Generally, updated firmware must be downloaded from the MultiTech web/FTP site to the user's PC before it can be downloaded from that PC to the MultiVOIP.

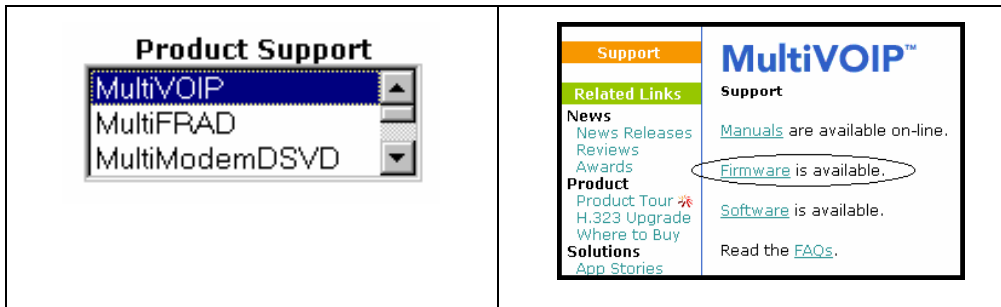
Note that the structure of the MultiTech web/FTP site may change without notice. However, firmware updates can generally be found using standard web techniques. For example, you can access updated firmware by doing a search or by clicking on **Support**.



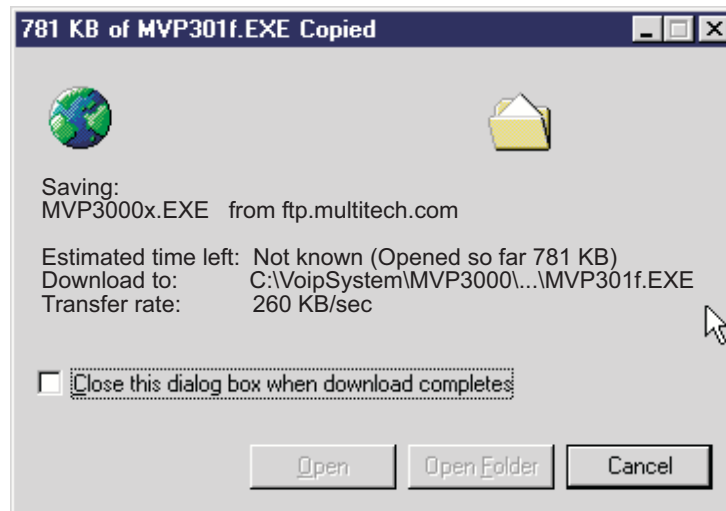
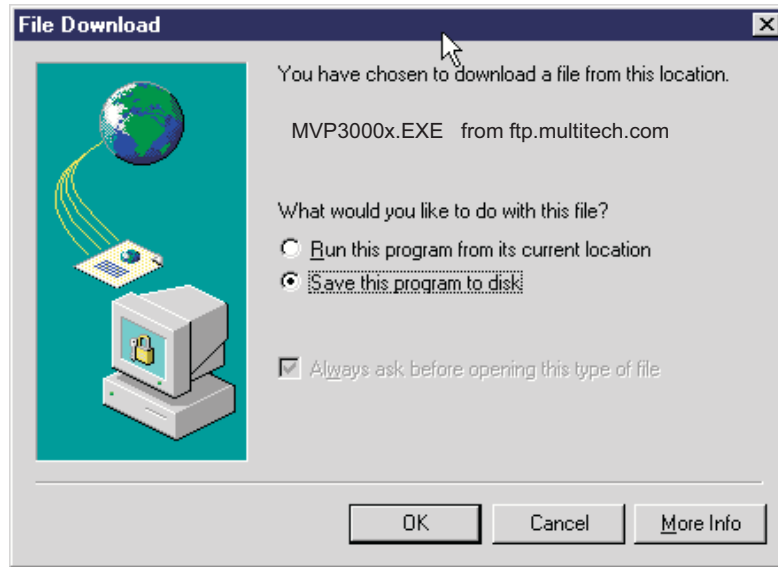
If you conduct a search, for example, on the word "MultiVoip," you will be directed to a list of firmware that can be downloaded.



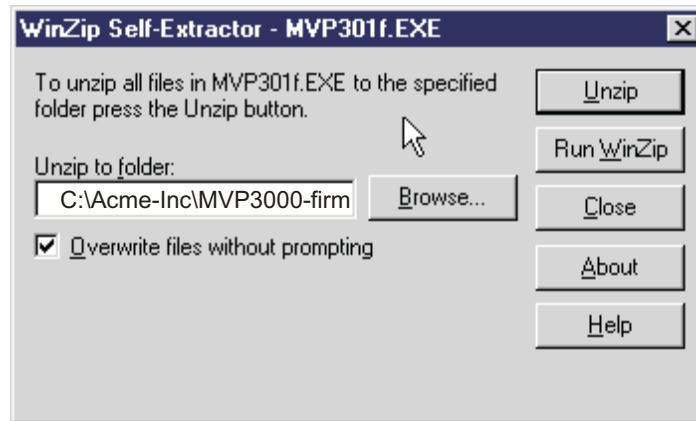
If you choose **Support**, you can select "MultiVoip" in the **Product Support** menu and then click on **Firmware** to find MultiVOIP resources.



Once the updated firmware has been located, it can be downloaded from the web/ftp site using normal PC/Windows procedures. While the next 3 screens below pertain to the MVP3010, similar screens will appear for any MultiVOIP model described in this manual.



Generally, the firmware file will be a self-extracting compressed file (with .zip extension), which must be expanded (decompressed, or “unzipped”) on the user’s PC in a user-specified directory.



## Upgrading MultiVOIP Firmware via FTP Client and Voip’s Built-In FTP Server Function

MultiTech has built an FTP server into the MultiVOIP unit. Therefore, file transfers from the controller PC to the voip unit can be done using an FTP client program or even using a browser (e.g., Internet Explorer, Netscape, or FireFox, used in conjunction with Windows Explorer).

The terminology of “downloads” and “uploads” gets a bit confusing in this context. File transfers from a client to a server are typically considered “uploads.” File transfers from a large repository of data to machines with less data capacity are considered “downloads.” In this case, these metaphors are contradictory: the FTP server is actually housed in the MultiVOIP unit, and the controller PC, which is actually the repository of the info to be transferred, uses an FTP client program. In this situation, we have chosen to call the transfer of files from the PC to the voip “downloads.” (Be aware that some FTP client programs may use the opposite terminology, i.e., they may refer to the file transfer as an “upload”)

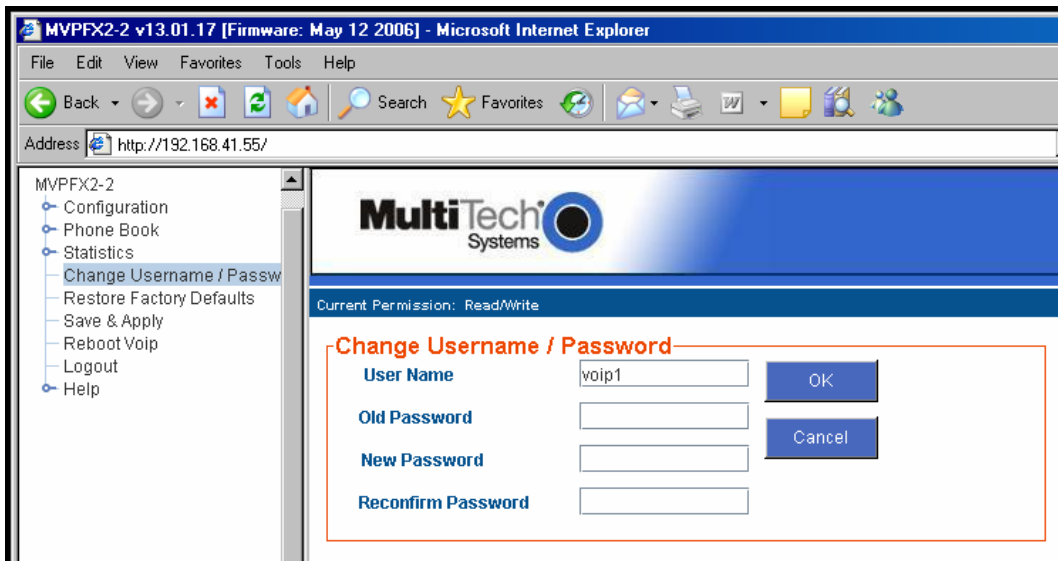
You can download firmware, the file system, and MSP firmware for the MultiVOIP unit with this FTP functionality. These downloads are done over a network, not by a local serial port connection. Consequently, voips at distant locations can be updated from a central control point.

To transfer files using the FTP server functionality in the MultiVOIP, follow these directions.

1. **Establish Network Connection and IP Addresses.** Both the controller PC and the MultiVOIP unit(s) must be connected to the same IP network. An IP address must be assigned for each.

IP Address of Control PC	_____ . _____ . _____ . _____
IP Address of voip unit #1	_____ . _____ . _____ . _____
:	:
:	:
IP address of voip unit #n	_____ . _____ . _____ . _____

2. **Establish User Name and Password.** You must establish a user name and (optionally) a password for contacting the voip over the IP network. (When connection is made via a local serial connection between the PC and the voip unit, no user name is needed.)

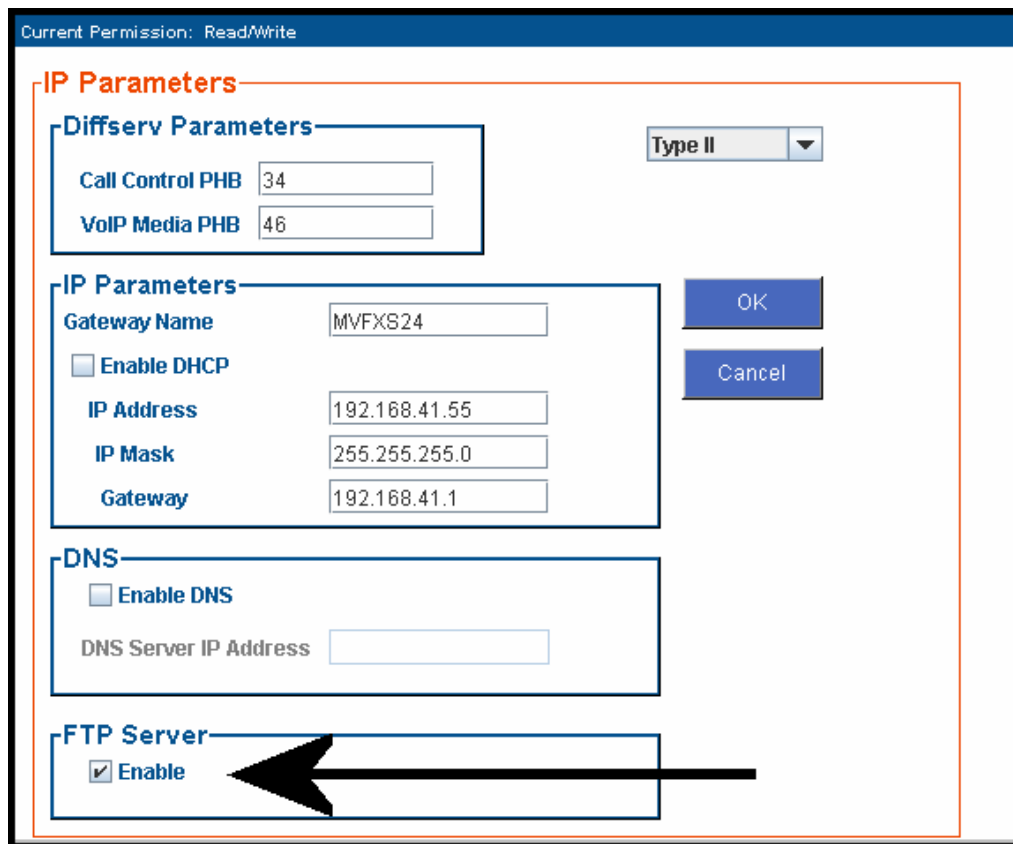


As shown above, the username and password must be set in the web GUI.

3. **Install FTP Client Program or Use Substitute.** You *should* install an FTP client program on the controller PC. FTP file transfers can be done using a web browser (e.g., Mozilla or Internet Explorer) in conjunction with a local Windows browser (e.g., Windows Explorer), but this approach is somewhat clumsy (it requires use of two application programs rather than one) and it limits downloading to only one VOIP unit at a time. With an FTP client program, multiple voips can receive FTP file transmissions in response to a single command (the transfers may occur serially however).

Although MultiTech does not provide an FTP client program with the MultiVOIP software or endorse any particular FTP client program, we remind our readers that adequate FTP programs are readily available under retail, shareware and freeware licenses. (Read and observe any End-User License Agreement carefully.) Two examples of this are the “WSFTP” client and the “SmartFTP” client, with the former having an essentially text-based interface and the latter having a more graphically oriented interface, as of this writing. User preferences will vary. Examples here show use of both programs.

4. **Enable FTP Functionality.** Go to the **IP Parameters** screen and click on the “FTP Server: Enable” box. FTP is enabled by default. You would need to re-enable it only if it had been turned off.



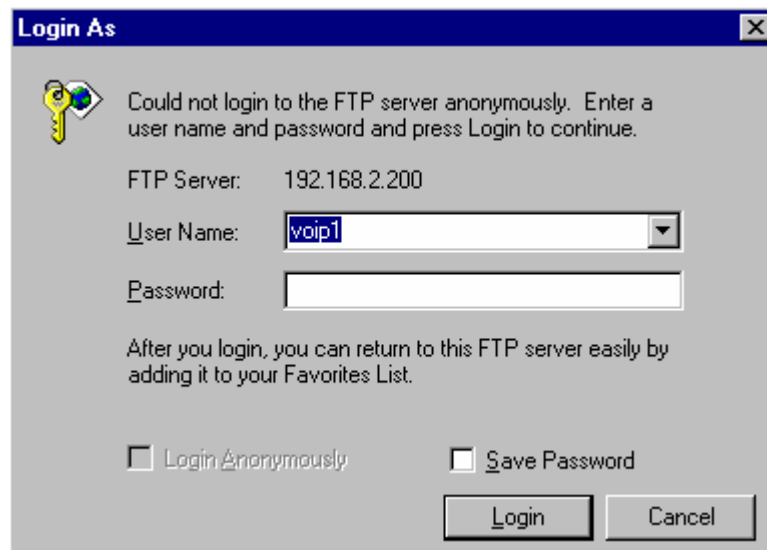
5. **Identify Files to be Updated.** Determine which files you want to update. Three types of files can be updated using the FTP feature.

File Type	File Names	Description
firmware "bin" file	mvpapp	This is the MultiVOIP firmware file. Only one file of this type will be in the directory.
file system	mvpfs-img	The MultiVOIP filesystem file in which are stored the GUI and the default values of GUI operating parameters.
MSP firmware	mstp-img	This is MSP firmware for the MultiVOIP's DSP processing unit.

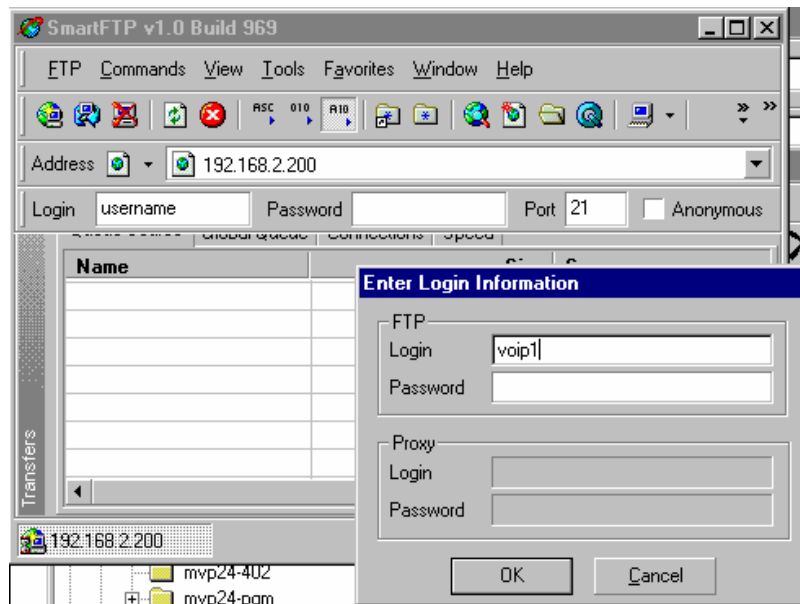
6. **Contact MultiVOIP FTP Server.** You must make contact with the FTP Server in the voip using either a web browser or FTP client program. Enter the IP address of the MultiVOIP's FTP Server. If you are using a browser, the address must be preceded by "ftp://" (otherwise you'll reach the web GUI within the MultiVOIP unit).



7. **Log In.** Use the User Name and password established in item #2 above. The login screens will differ depending on whether the FTP file transfer is to be done with a web browser (see first screen below) or with an FTP client program (see second screen below).





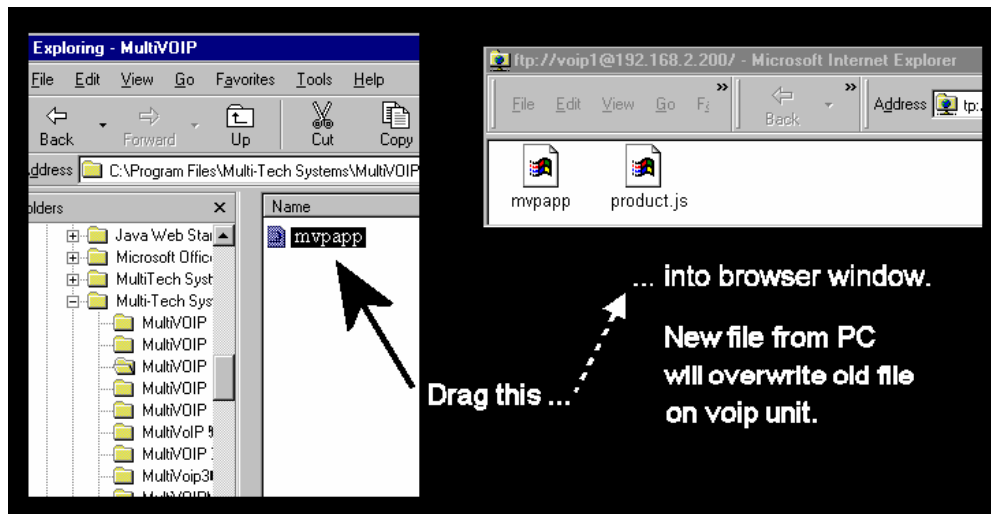


**8. Invoke Download.** Downloading can be done with a web browser or with an FTP client program.

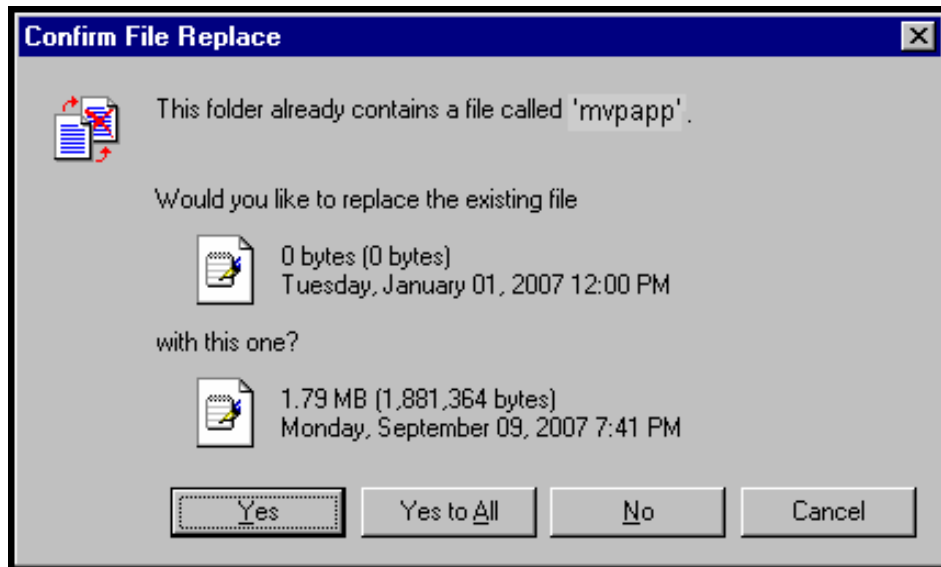
**8A. Download with Web Browser.**

8A1. In the local Windows browser, locate the directory holding the MultiVOIP program files. The default location will be C:\Program Files \Multi-Tech Systems \MultiVOIP xxxx yyyy (where x and y represent MultiVOIP model numbers and software version numbers).

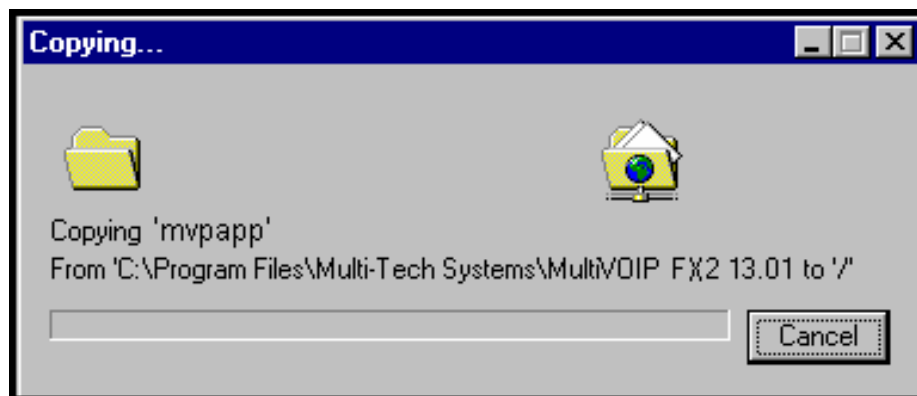
8A2. Drag-and-drop files from the local Windows browser (e.g., Windows Explorer) to the web browser.



You may be asked to confirm the overwriting of files on the MultiVOIP. Do so.



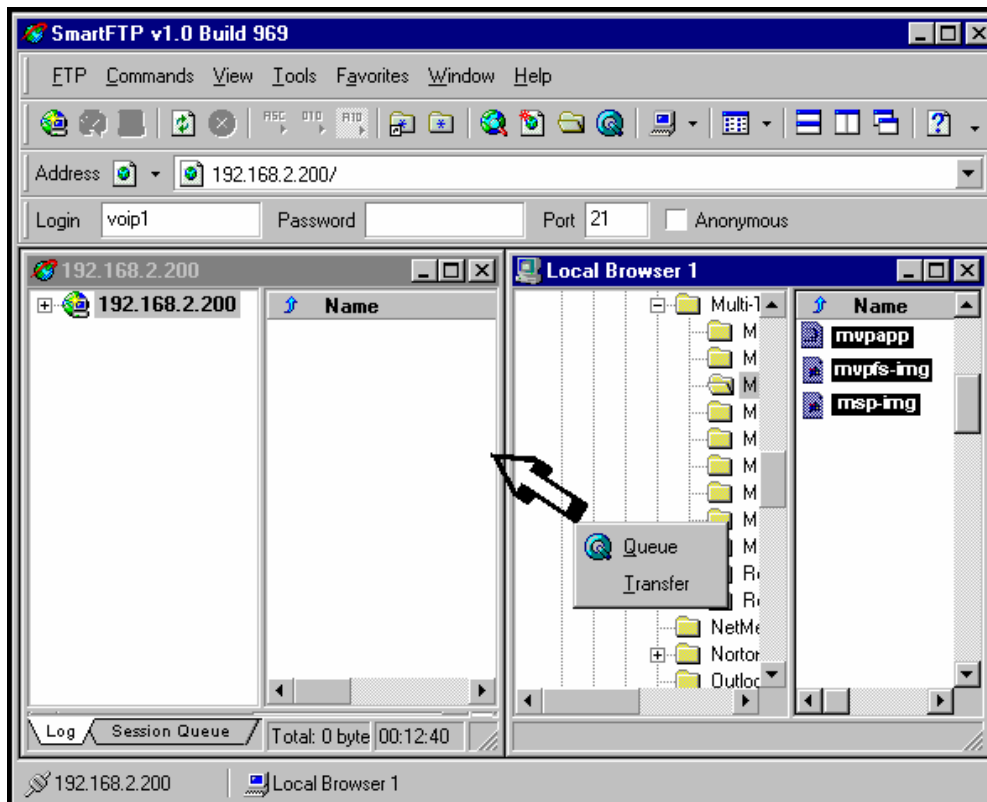
File transfer between PC and voip will look like transfer within voip directories.



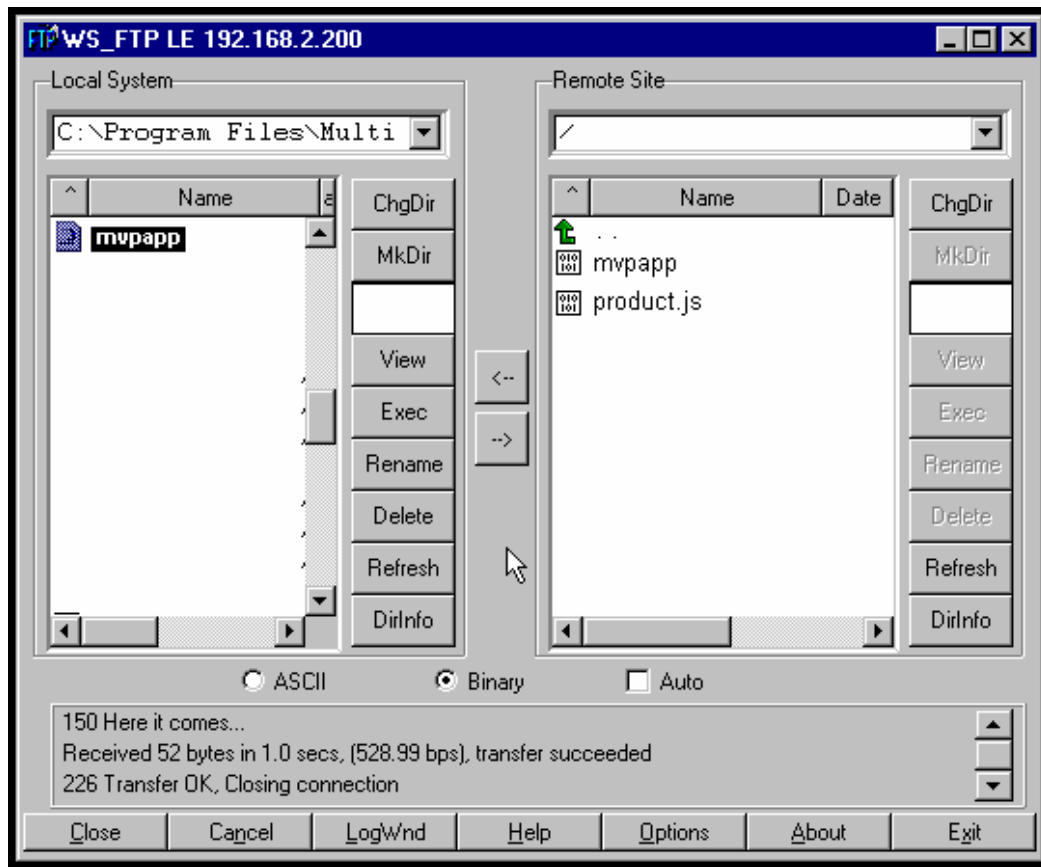
Before attempting to update any other files, wait at least 3 minutes (which allows time for the MultiVOIP to reboot). (When the **Boot** light is off, the rebooting process is complete.) After this 3-minute wait, you can re-connect to the MultiVOIP's FTP server and update another file.

## 8B. Download with FTP Client Program.

- 8B1. In the local directory browser of the FTP client program, locate the directory holding the MultiVOIP program files. The default location will be C:\Program Files \Multi-Tech Systems \MultiVOIP xxxx yyyy (where x and y represent MultiVOIP model numbers and software version numbers).
- 8B2. In the FTP client program window, drag-and-drop files from the local browser pane to the pane for the MultiVOIP FTP server. FTP client GUI operations vary. In some cases, you can choose between immediate and queued transfer. In some cases, there may be automated capabilities to transfer to multiple destinations with a single command.



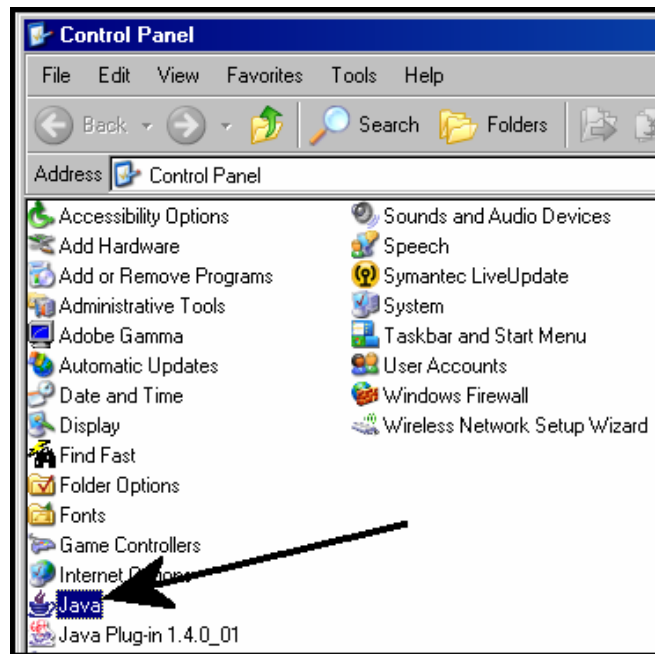
Some FTP client programs are more graphically oriented (see previous screen), while others (like the “WS-FTP” client) are more text oriented.



9. Before attempting to update any other files, wait at least 3 minutes (which allows time for the MultiVOIP to reboot). (When the **Boot** light is off, the rebooting process is complete.) After this 3-minute wait, you can re-connect to the MultiVOIP’s FTP server and update another file.

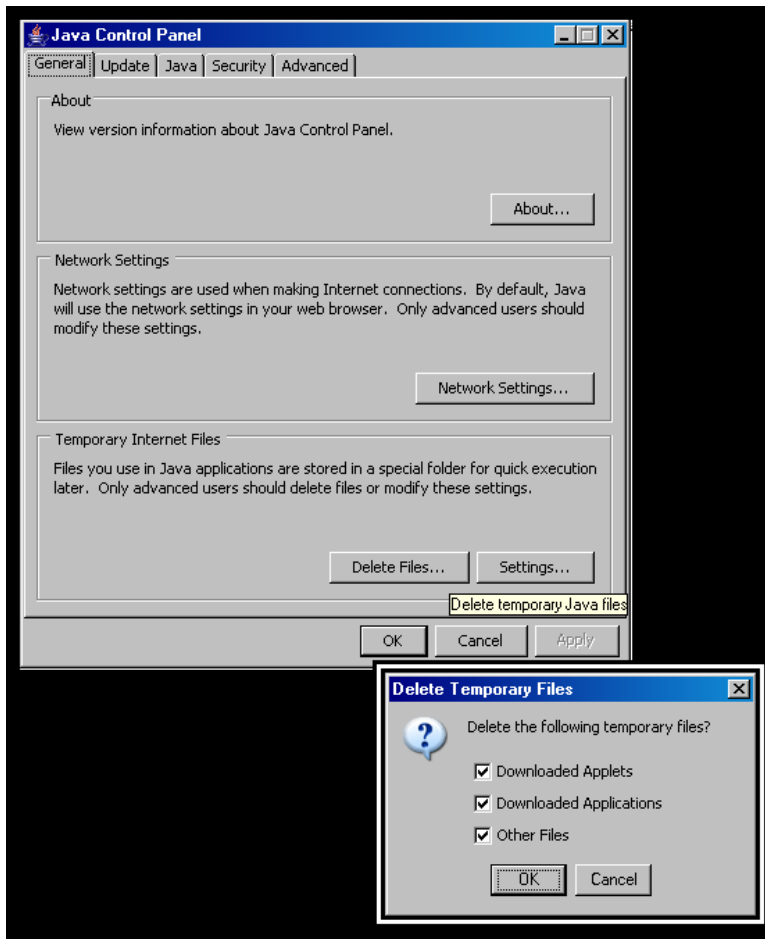
10. Browse to the MultiVOIP’s web interface and look at the **Configuration | System Information** screen to confirm that the firmware has been updated to the appropriate version.

11. After updating the MultiVOIP's file system image file (mvpfs-img), you must clear the Java cache of files that pertain to the old/outdated version of the image file.
  - A. To clear the Java cache in Windows, go to **Start | Settings | Control Panel**. Select **Java** in the list.



- B. In the **General** tab of the Java program, click "Delete Files."

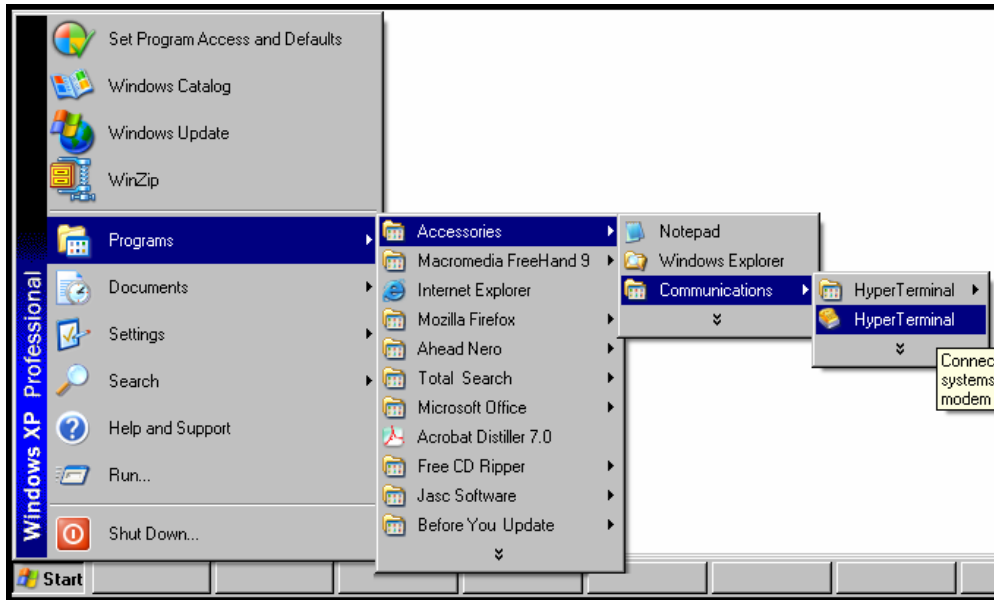
C. When the **Delete Temporary Files** screen appears, click **OK**. Then click **OK** again at the main Java screen.



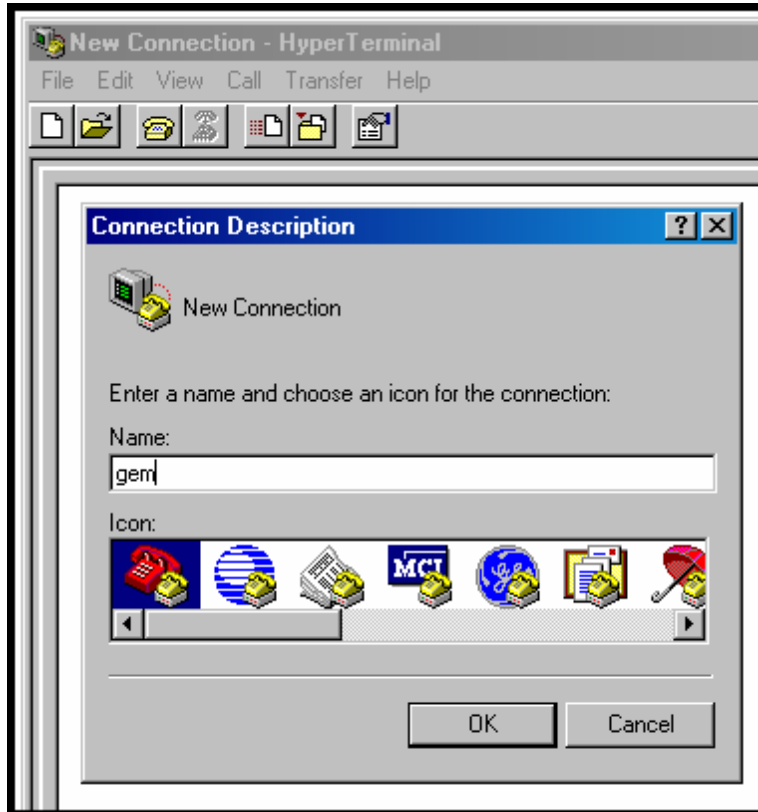
D. Using a new browser window, go to the IP address of the MultiVOIP. All of the new features of the updated file system image file will now be visible.

## Upgrading MultiVOIP Firmware via TFTP using HyperTerminal

1. Before beginning this procedure, you must have a TFTP server program running on a computer that has access to the network on which the MultiVOIP is running and the upgrade software files must be on that computer. TFTP server programs can be downloaded for free from various Internet web sites.
2. Connect a cable between the MultiVOIP's "Console" connector and a serial cable on the computer.
3. Launch HyperTerminal or a similar communications program.



4. Establish a 'connection' in HyperTerminal.

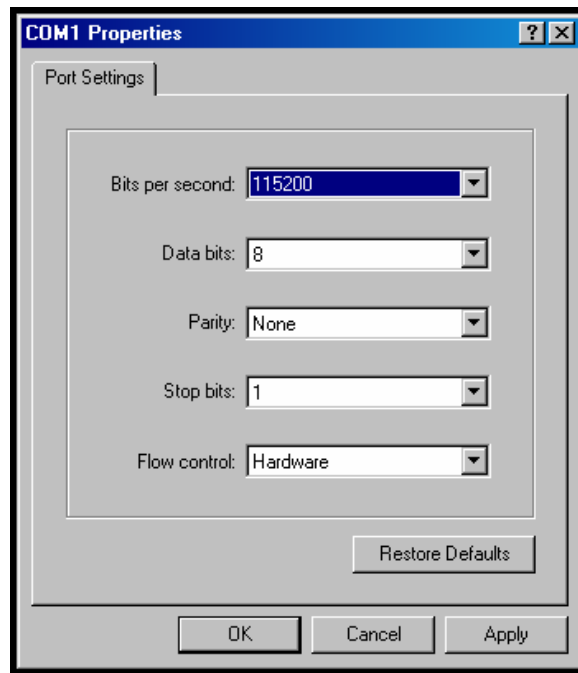


5. Check that HyperTerminal is addressing the correct COM port.

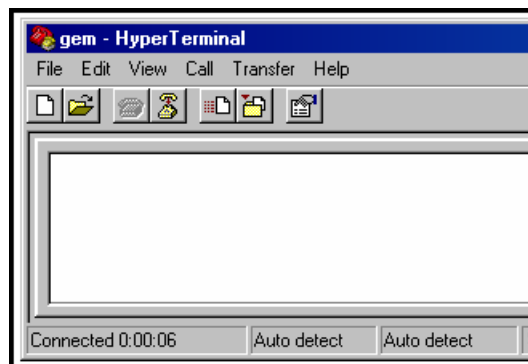




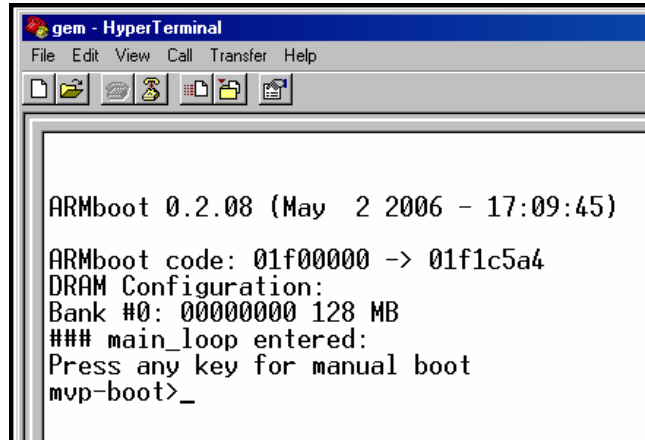
6. Check that HyperTerminal's data rate is set to 115200bps.



7. To begin, HyperTerminal must be connected and ready.



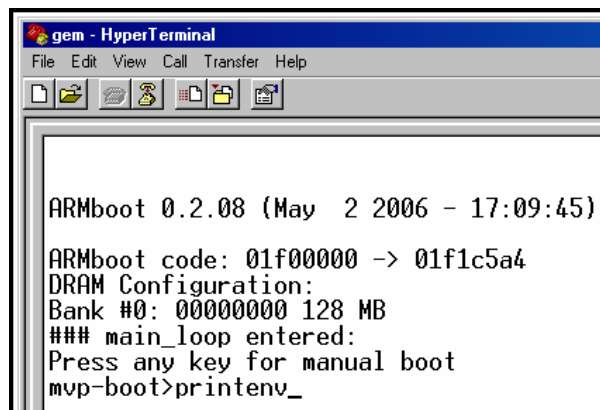
8. Reboot the MultiVOIP by turning off its power and turning it back on again. The ARMBoot prompt will appear on the HyperTerminal screen.



```
gem - HyperTerminal
File Edit View Call Transfer Help
ARMboot 0.2.08 (May 2 2006 - 17:09:45)
ARMboot code: 01f00000 -> 01f1c5a4
DRAM Configuration:
Bank #0: 00000000 128 MB
### main_loop entered:
Press any key for manual boot
mvp-boot>_
```

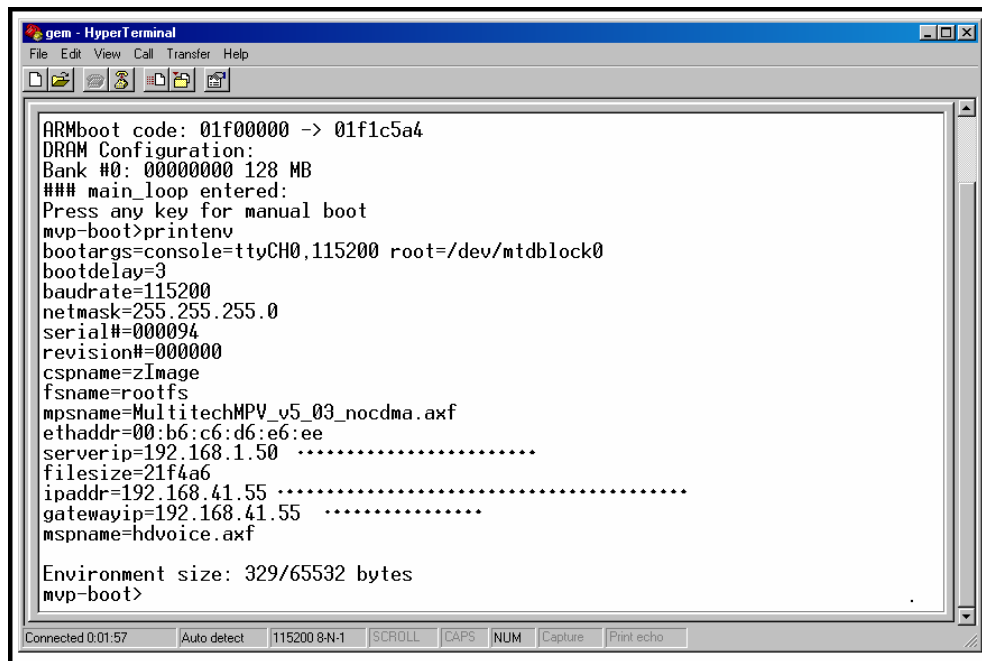
When this screen appears, you must quickly press any key to stop the regular boot-up process (the manual boot process).

9. To view voip parameters, type **printenv** at the **mvp-boot>** prompt. Then press **Enter**.



```
gem - HyperTerminal
File Edit View Call Transfer Help
ARMboot 0.2.08 (May 2 2006 - 17:09:45)
ARMboot code: 01f00000 -> 01f1c5a4
DRAM Configuration:
Bank #0: 00000000 128 MB
### main_loop entered:
Press any key for manual boot
mvp-boot>printenv_
```

10. A list of voip parameters that can be altered in the ARMBoot environment will appear.



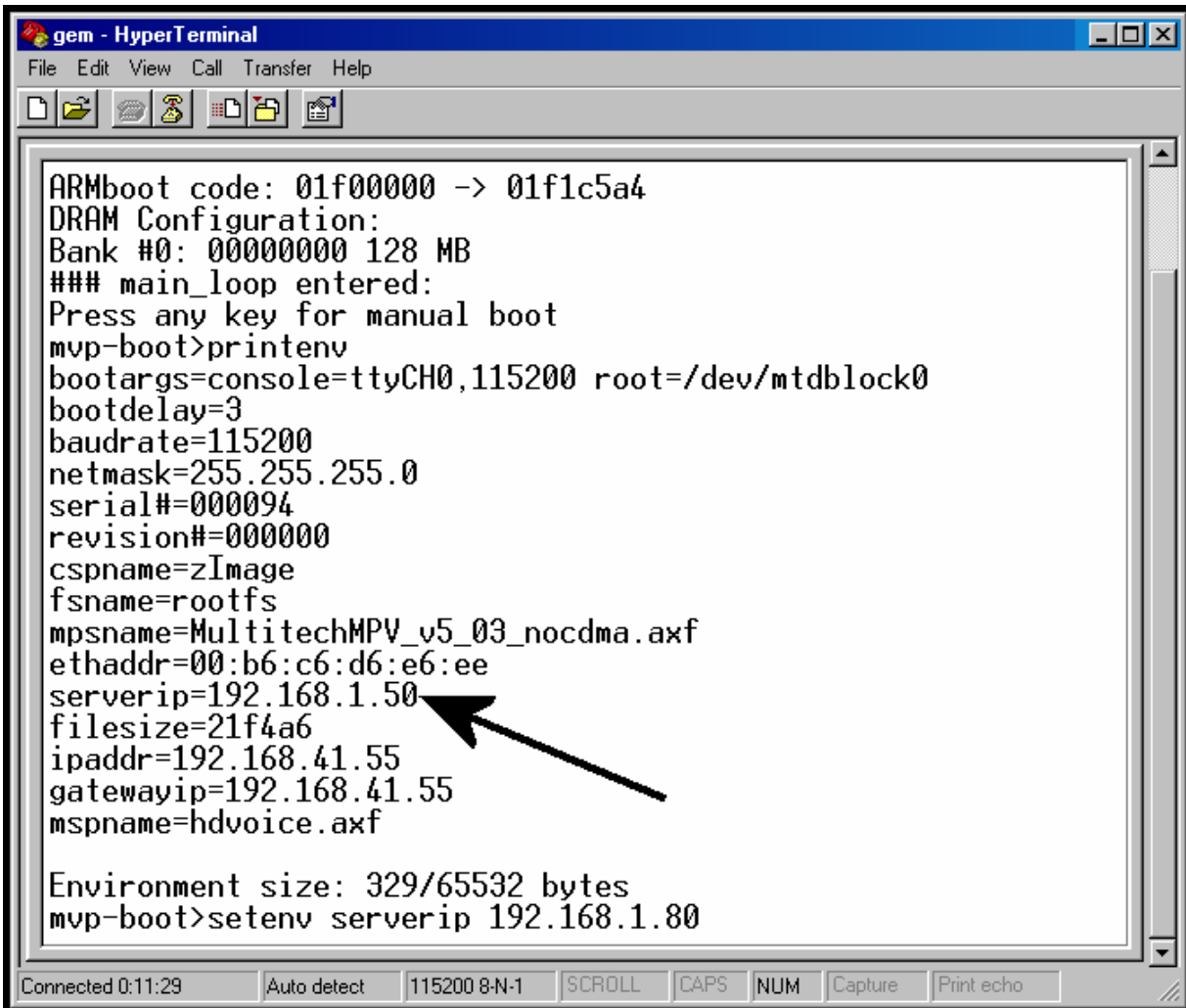
The image shows a HyperTerminal window titled "gem - HyperTerminal". The window displays the output of an ARMBoot environment. The text shown is as follows:

```
ARMboot code: 01f00000 -> 01f1c5a4
DRAM Configuration:
Bank #0: 00000000 128 MB
### main_loop entered:
Press any key for manual boot
mvp-boot>printenv
bootargs=console=ttyCH0,115200 root=/dev/mtdblock0
bootdelay=3
baudrate=115200
netmask=255.255.255.0
serial#=000094
revision#=000000
cspname=zImage
fsname=rootfs
mpsname=MultitechMPV_v5_03_nocdma.axf
ethaddr=00:b6:c6:d6:e6:ee
serverip=192.168.1.50 .....
filesize=21f4a6
ipaddr=192.168.41.55 .....
gatewayip=192.168.41.55 .....
mspname=hdvoice.axf

Environment size: 329/65532 bytes
mvp-boot>
```

The status bar at the bottom of the window shows "Connected 0:01:57", "Auto detect", "115200 8-N-1", "SCROLL", "CAPS", "NUM", "Capture", and "Print echo".

11. You must change the **serverip** value to the IP address of the computer on which the TFTP server program is located. To change the serverip value, type **setenv serverip a.b.c.d** (where a, b, c, and d are the four octet values for the IP address of the TFTP server) at the **mvp-boot>** prompt. Then press **Enter**.

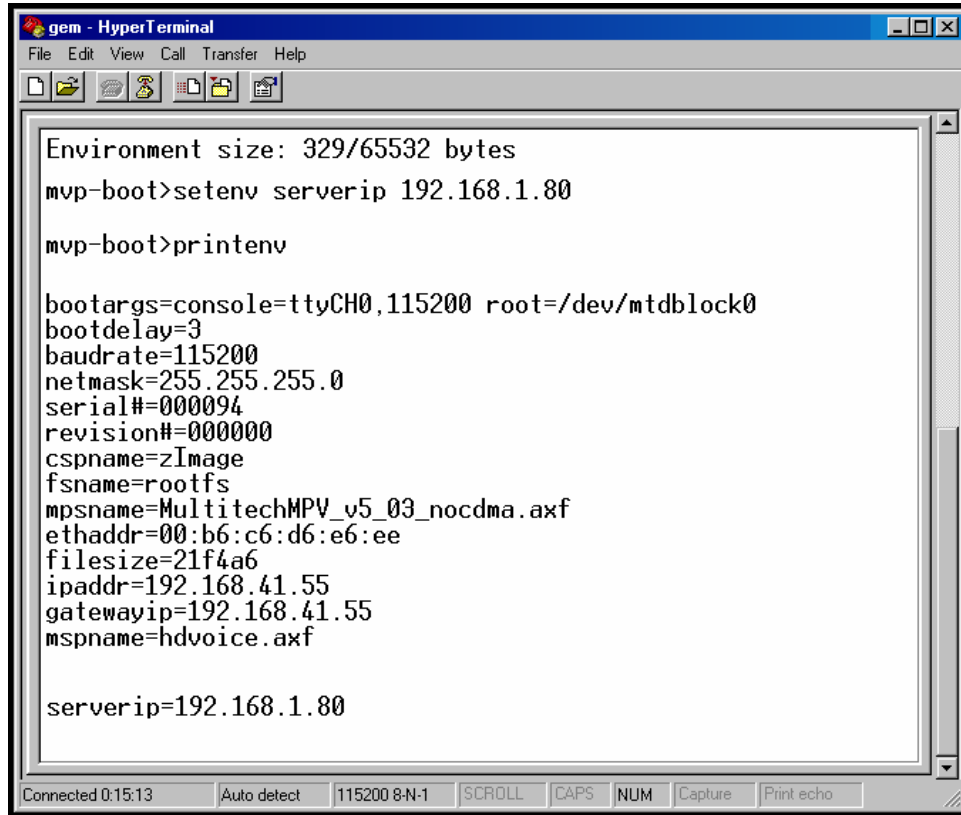


```
gem - HyperTerminal
File Edit View Call Transfer Help
ARMboot code: 01f00000 -> 01f1c5a4
DRAM Configuration:
Bank #0: 00000000 128 MB
### main_loop entered:
Press any key for manual boot
mvp-boot>printenv
bootargs=console=ttyCH0,115200 root=/dev/mtdblock0
bootdelay=3
baudrate=115200
netmask=255.255.255.0
serial#=000094
revision#=000000
cspname=zImage
fsname=rootfs
mpsname=MultitechMPV_v5_03_nocdma.axf
ethaddr=00:b6:c6:d6:e6:ee
serverip=192.168.1.50
filesize=21f4a6
ipaddr=192.168.41.55
gatewayip=192.168.41.55
mpsname=hdvoice.axf

Environment size: 329/65532 bytes
mvp-boot>setenv serverip 192.168.1.80
```

Note: When using the **setenv** command, be careful in your spelling. If you mis-spell **serverip** as “seeverip” for example, the ARMBoot program will create a new and useless variable entitled **seeverip** and will not change the value of the **serverip** variable.

12. To confirm that the TFTP server IP address was indeed changed to the value you want, type **printenv** at the **mvp-boot>** prompt and then press **Enter**.



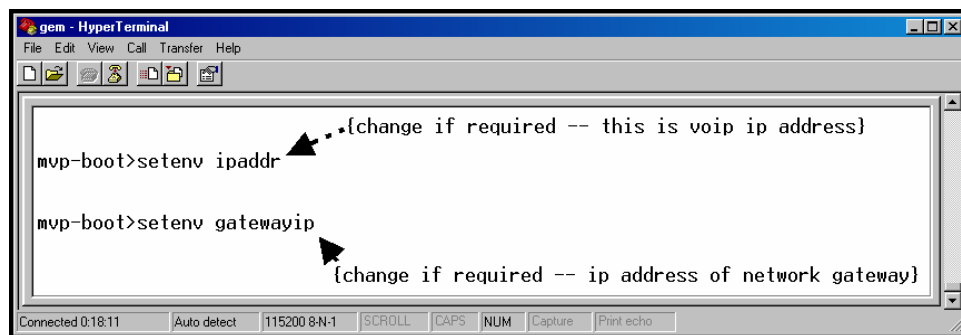
```

gem - HyperTerminal
File Edit View Call Transfer Help
Environment size: 329/65532 bytes
mvp-boot>setenv serverip 192.168.1.80
mvp-boot>printenv
bootargs=console=ttyCH0,115200 root=/dev/mtdblock0
bootdelay=3
baudrate=115200
netmask=255.255.255.0
serial#=000094
revision#=000000
cspname=zImage
fsname=rootfs
mpsname=MultitechMPV_v5_03_nocdma.axf
ethaddr=00:b6:c6:d6:e6:ee
filesize=21f4a6
ipaddr=192.168.41.55
gatewayip=192.168.41.55
mspname=hdvoice.axf

serverip=192.168.1.80

```

13. If necessary, you can also change the IP address of the voip (which is the **ipaddress** field in the ARMBoot environment) and the IP address of the network gateway (which is the **gatewayip** field in the ARMBoot environment)



```

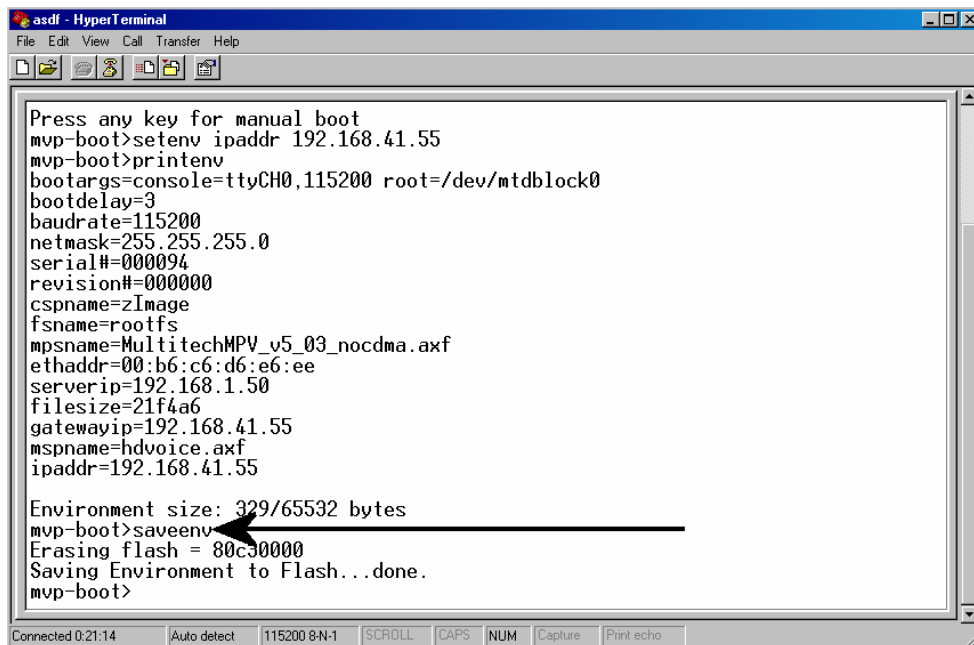
gem - HyperTerminal
File Edit View Call Transfer Help
mvp-boot>setenv ipaddr [change if required -- this is voip ip address]
mvp-boot>setenv gatewayip [change if required -- ip address of network gateway]

```

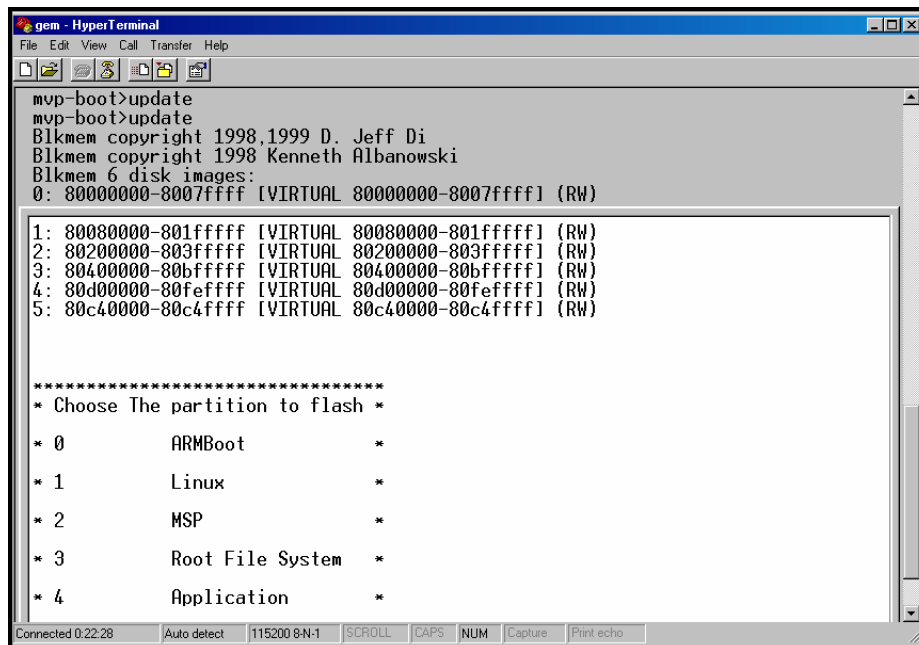
For **ipaddr**, type **setenv ipaddr a.b.c.d** (where a, b, c, and d are the four octet values for the IP address of the TFTP server) at the **mvp-boot>** prompt. Then press **Enter**.

For **gatewayip**, type **setenv gatewayip a.b.c.d** (where a, b, c, and d are the four octet values for the IP address of the TFTP server) at the **mvp-boot>** prompt. Then press **Enter**.

14. . Type **saveenv** and press Enter.



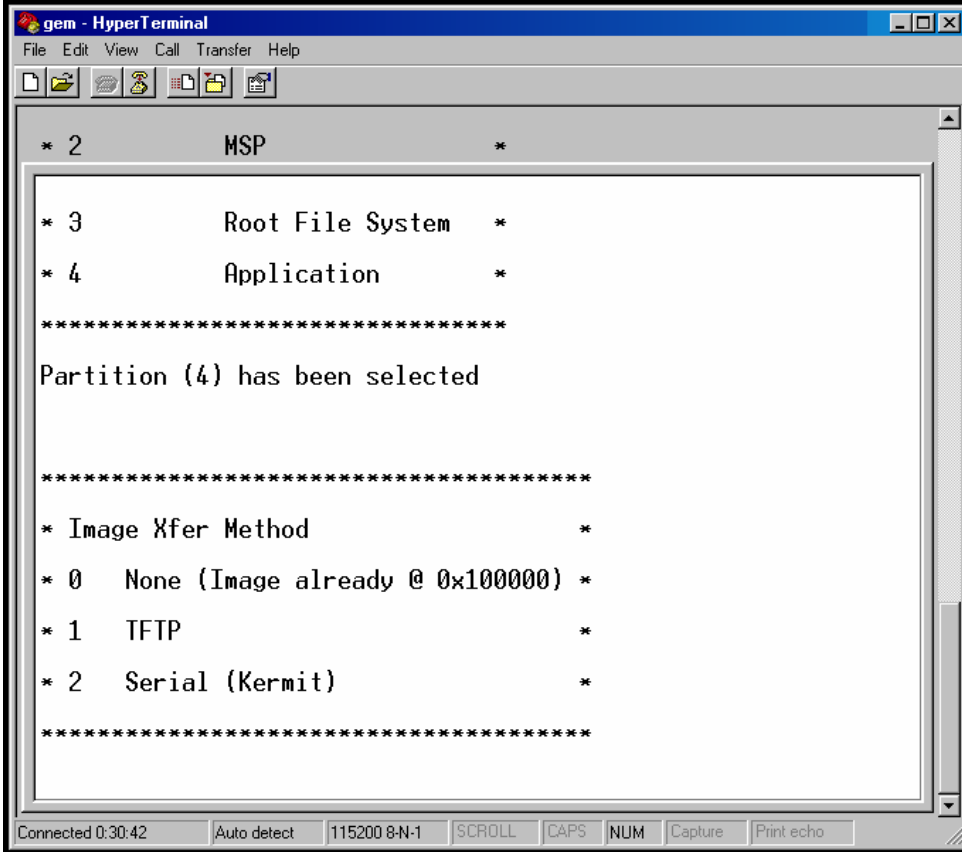
15. At the **mvp-boot>** prompt, type **update** and then press **Enter**.



A menu will appear that lists, by number, the various firmware entities (“partitions”) that could be updated with this command. When you choose an item from this list, the update for that firmware entity will begin as soon as you enter the number (the ‘application-update’ command is invoked without pressing **Enter**).

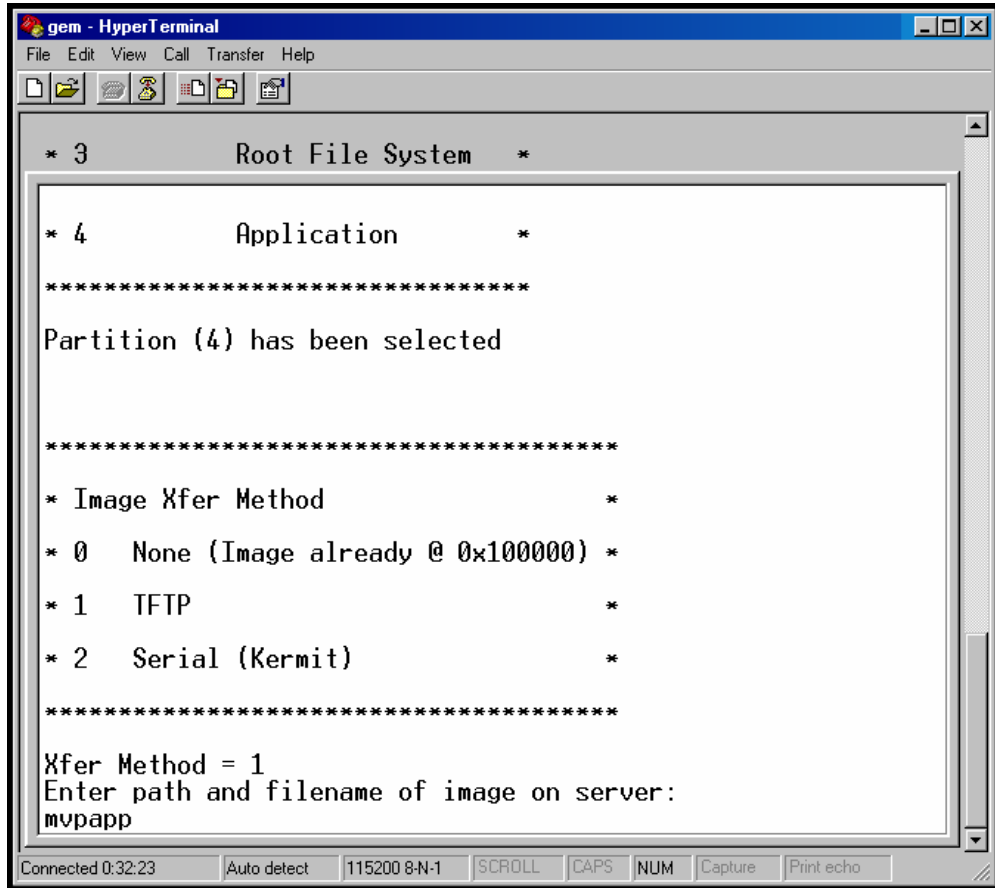
16. At the **mvp-boot>** prompt, type **4** to update the application.

**NOTE:** The file system (the appropriate file name is “mvpfs-img” with no file extension) can also be updated at this menu by typing **3** . The MSP firmware can be updated at this menu by typing **2** .



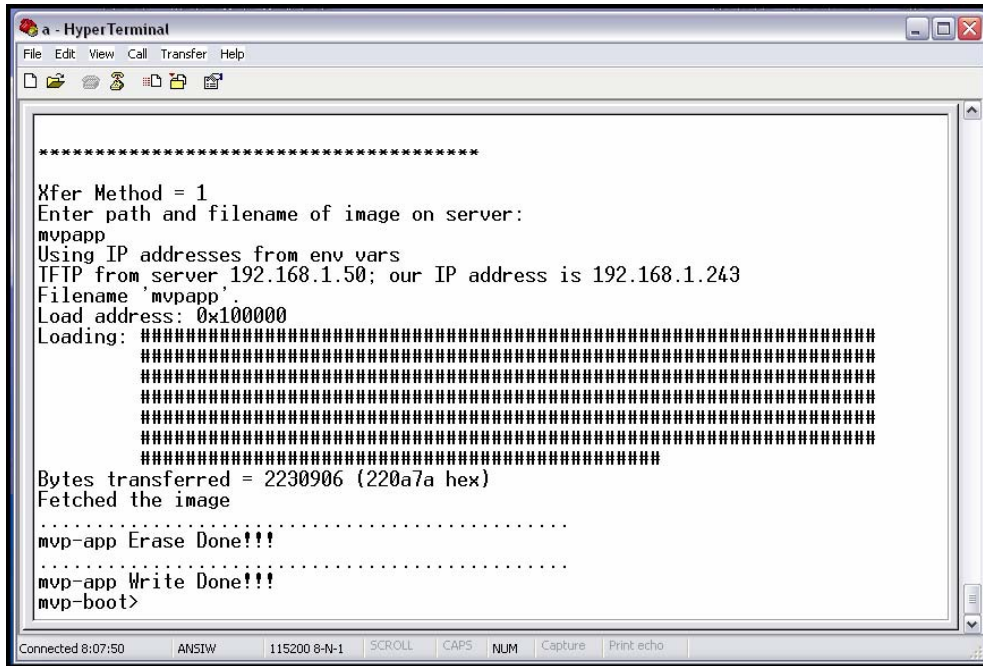
```
gem - HyperTerminal
File Edit View Call Transfer Help
* 2      MSP      *
* 3      Root File System  *
* 4      Application      *
*****
Partition (4) has been selected
*****
* Image Xfer Method      *
* 0  None (Image already @ 0x100000) *
* 1  TFTP                *
* 2  Serial (Kermit)     *
*****
Connected 0:30:42  Auto detect  115200 8-N-1  SCROLL  CAPS  NUM  Capture  Print echo
```

17. When the **Image Xfer Method** menu appears, type **1** . (Option 2 will also work, but it could take as long as 45 minutes to accomplish the transfer by using the “Serial - Kermit” connection.)





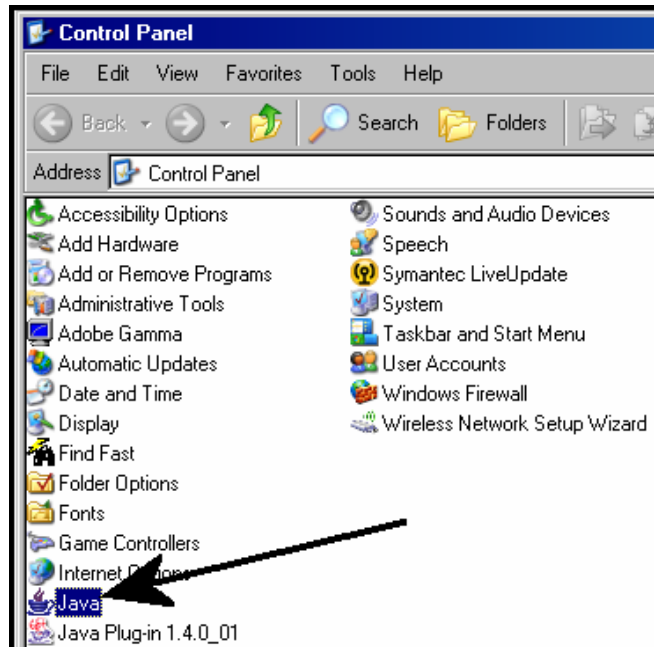
18. The transfer process will take a few minutes. When complete, the response “Write done !!!” will appear.



After the “Write done!!!” message appears, you can then update other firmware partitions (like the file system and the MSP firmware) using the steps presented above in this procedure. When you are done updating all of the files that need updating, reboot the voip by turning its power off and back on.

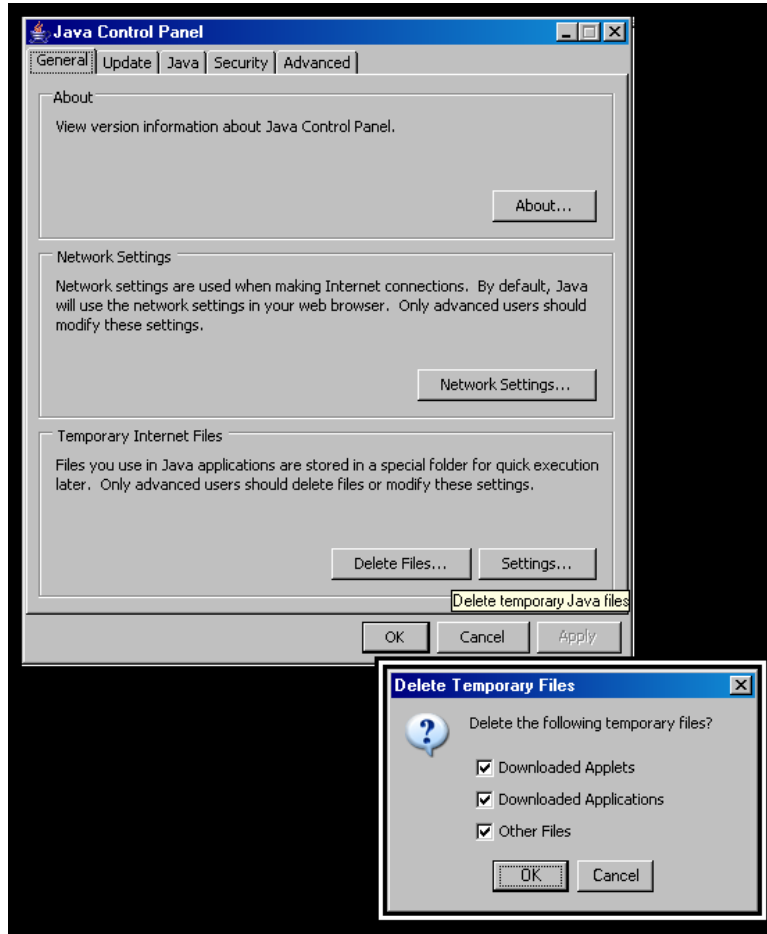
19. After updating the MultiVOIP’s file system image file (mvpfs-img), you must clear the Java cache of files that pertain to the old/outdated version of the image file.

- A. To clear the Java cache in Windows, go to **Start | Settings | Control Panel**. Select **Java** in the list.



B. In the **General** tab of the Java program, click “Delete Files.”

C. When the **Delete Temporary Files** screen appears, click **OK**. Then click **OK** again at the main Java screen.



D. Using a new browser window, go to the IP address of the MultiVOIP. All of the new features of the updated file system image file will now be visible.

## SysLog Server Functions

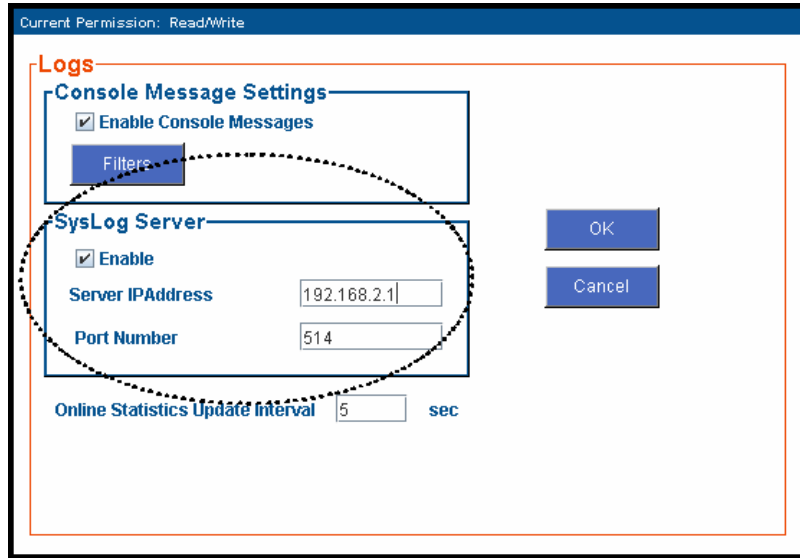
MultiTech has built SysLog server functionality into the software of the MultiVOIP units. SysLog is a *de facto* standard for logging events in network communication systems.

The SysLog Server resides in the MultiVOIP unit itself. To implement this functionality, you will need a SysLog client program (sometimes referred to as a “daemon”). SysLog client programs, both paid and freeware, can be obtained from Kiwi Enterprises, among other firms. Read the End-User License Agreement carefully and observe license requirements. See [www.kiwisyslog.com](http://www.kiwisyslog.com). SysLog client programs essentially give you a means of structuring console messages for convenience and ease of use.

MultiTech Systems does not endorse any particular SysLog client program. SysLog client programs by qualified providers should suffice for use with MultiVOIP units. Kiwi’s brief description of their SysLog program is as follows:

“Kiwi Syslog Daemon is a freeware Syslog Daemon for the Windows platform. It receives, logs, displays and forwards Syslog messages from hosts such as routers, switches, Unix hosts and any other syslog enabled device. There are many customizable options available.”

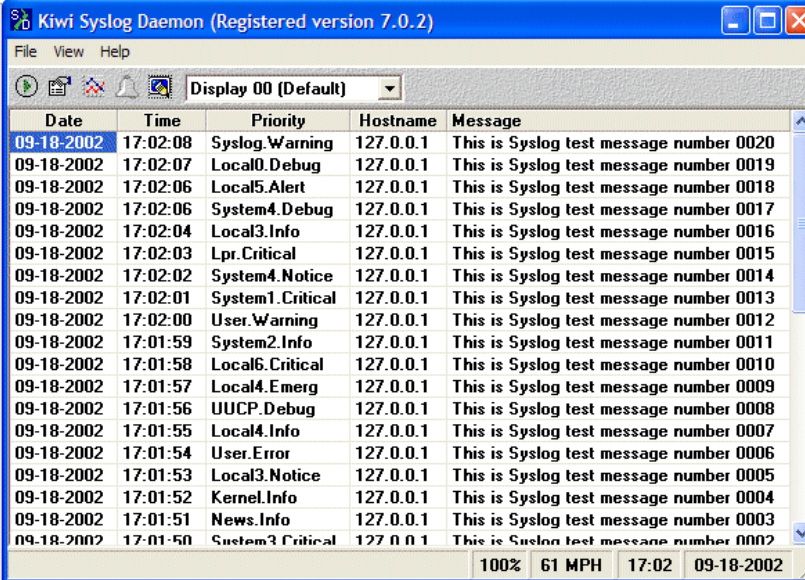
Before a SysLog client program is used, the SysLog functionality must be enabled within the MultiVOIP in the **Logs** menu under **Configuration**.



The IP Address used will be that of the MultiVOIP itself.

In the **Port** field, entered by default, is the standard ('well-known') logical port, 514.

**Configuring the SysLog Client Program.** Configure the SysLog client program for your own needs. In various SysLog client programs, you can define where log messages will be saved/archived, opt for interaction with an SNMP system (not applicable for MVPFX2 units), set the content and format of log messages, determine disk space allocation limits for log messages, and establish a hierarchy for the seriousness of messages (normal, alert, critical, emergency, etc.). A sample presentation of SysLog info in the Kiwi daemon is shown below. SysLog programs will vary in features and presentation.



The screenshot shows the 'Kiwi Syslog Daemon (Registered version 7.0.2)' application window. It features a menu bar with 'File', 'View', and 'Help'. Below the menu bar is a toolbar with several icons and a dropdown menu set to 'Display 00 (Default)'. The main area contains a table of log messages with the following columns: Date, Time, Priority, Hostname, and Message. The messages are sorted by time, showing various priority levels such as Warning, Alert, Debug, Info, Critical, and Notice. At the bottom of the window, there is a status bar displaying '100%', '61 MPH', '17:02', and '09-18-2002'.

Date	Time	Priority	Hostname	Message
09-18-2002	17:02:08	Syslog.Warning	127.0.0.1	This is Syslog test message number 0020
09-18-2002	17:02:07	Local0.Debug	127.0.0.1	This is Syslog test message number 0019
09-18-2002	17:02:06	Local5.Alert	127.0.0.1	This is Syslog test message number 0018
09-18-2002	17:02:06	System4.Debug	127.0.0.1	This is Syslog test message number 0017
09-18-2002	17:02:04	Local3.Info	127.0.0.1	This is Syslog test message number 0016
09-18-2002	17:02:03	Lpr.Critical	127.0.0.1	This is Syslog test message number 0015
09-18-2002	17:02:02	System4.Notice	127.0.0.1	This is Syslog test message number 0014
09-18-2002	17:02:01	System1.Critical	127.0.0.1	This is Syslog test message number 0013
09-18-2002	17:02:00	User.Warning	127.0.0.1	This is Syslog test message number 0012
09-18-2002	17:01:59	System2.Info	127.0.0.1	This is Syslog test message number 0011
09-18-2002	17:01:58	Local6.Critical	127.0.0.1	This is Syslog test message number 0010
09-18-2002	17:01:57	Local4.Emerg	127.0.0.1	This is Syslog test message number 0009
09-18-2002	17:01:56	UUCP.Debug	127.0.0.1	This is Syslog test message number 0008
09-18-2002	17:01:55	Local4.Info	127.0.0.1	This is Syslog test message number 0007
09-18-2002	17:01:54	User.Error	127.0.0.1	This is Syslog test message number 0006
09-18-2002	17:01:53	Local3.Notice	127.0.0.1	This is Syslog test message number 0005
09-18-2002	17:01:52	Kernel.Info	127.0.0.1	This is Syslog test message number 0004
09-18-2002	17:01:51	News.Info	127.0.0.1	This is Syslog test message number 0003
09-18-2002	17:01:50	System3.Critical	127.0.0.1	This is Syslog test message number 0002

---

## **Chapter 8 Warranty, Service, and Tech Support**

## Limited Warranty

Multi-Tech Systems, Inc. ("MTS") warrants that its products will be free from defects in material or workmanship for a period of two years from the date of purchase, or if proof of purchase is not provided, two years from date of shipment. MTS MAKES NO OTHER WARRANTY, EXPRESSED OR IMPLIED, AND ALL IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE HEREBY DISCLAIMED. This warranty does not apply to any products which have been damaged by lightning storms, water, or power surges or which have been neglected, altered, abused, used for a purpose other than the one for which they were manufactured, repaired by the customer or any party without MTS's written authorization, or used in any manner inconsistent with MTS's instructions.

MTS's entire obligation under this warranty shall be limited (at MTS's option) to repair or replacement of any products which prove to be defective within the warranty period, or, at MTS's option, issuance of a refund of the purchase price. Defective products must be returned by Customer to MTS's factory – transportation prepaid.

MTS WILL NOT BE LIABLE FOR CONSEQUENTIAL DAMAGES AND UNDER NO CIRCUMSTANCES WILL ITS LIABILITY EXCEED THE PURCHASE PRICE FOR DEFECTIVE PRODUCTS.

## Repair Procedures for U.S. and Canadian Customers

In the event that service is required, products may be shipped, freight prepaid, to our Mounds View, Minnesota factory:

Multi-Tech Systems, Inc.  
2205 Woodale Drive  
Mounds View, MN 55112  
Attn: Repairs, Serial # \_\_\_\_\_

A Returned Materials Authorization (RMA) is not required. Return shipping charges (surface) will be paid by MTS.

Please include, inside the shipping box, a description of the problem, a return shipping address (it must be a street address, not a P.O. Box number), your telephone number, and if the product is out of warranty, a check or purchase order for repair charges.

For out-of-warranty repair charges, go to [www.multitech.com/documents/warranties](http://www.multitech.com/documents/warranties)

Extended two-year overnight replacement service agreements are available for selected products. Please call MTS at (888) 288-5470, extension 5308, or visit our web site at [www.multitech.com/programs/orc](http://www.multitech.com/programs/orc) for details on rates and coverages.

Please direct your questions regarding technical matters, product configuration, verification that the product is defective, etc., to our Technical Support department at (800) 972-2439 or email [tsupport@multitech.com](mailto:tsupport@multitech.com). Please direct your questions regarding repair expediting, receiving, shipping, billing, etc., to our Repair Accounting department at (800) 328-9717 or (763) 717-5631, or email [mtsrepair@multitech.com](mailto:mtsrepair@multitech.com).

Repairs for damages caused by lightning storms, water, power surges, incorrect installation, physical abuse, or used-caused damages are billed on a time-plus-materials basis.

## Technical Support

Multi-Tech Systems has an excellent staff of technical support personnel available to help you get the most out of your Multi-Tech product. If you have any questions about the operation of this unit, or experience difficulty during installation you can contact Tech Support via the following:

### Contacting Technical Support

Country	By E-mail	By telephone
France	<a href="mailto:support@multitech.fr">support@multitech.fr</a>	(33) 1-64 61 09 81
India	support@ multitechindia.com	(91) 124-340778
U.K.	support@ multitech.co.uk	(44) 118 959 7774
U.S. & Canada	tsupport@ multitech.com	(800) 972-2439
Rest of World	support@ multitech.com	(763) 785-3500

Internet: [http://www.multitech.com/\\_forms/email\\_tech\\_support.htm](http://www.multitech.com/_forms/email_tech_support.htm)

Please have your product information available, including model and serial number.



---

## **Chapter 9: Regulatory Information**



## EMC, Safety, and R&TTE Directive Compliance

The CE mark is affixed to this product to confirm compliance with the following European Community Directives:

Council Directive 89/336/EEC of 3 May 1989 on the approximation of the laws of Member States relating to electromagnetic compatibility,  
and

Council Directive 73/23/EEC of 19 February 1973 on the harmonization of the laws of Member States relating to electrical equipment designed for use within certain voltage limits,  
and

Council Directive 1999/5/EC of 9 March 1999 on radio equipment and telecommunications terminal equipment and the mutual recognition of their conformity.

## FCC Declaration

**NOTE:** This equipment has been tested and found to comply with the limits for a **Class A** digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy, and if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

This device complies with Part 15 of the FCC rules.

Operation is subject to the following two conditions:

- (1) This device may not cause harmful interference.
- (2) This device must accept any interference that may cause undesired operation.

**Warning:** Changes or modifications to this unit not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

## Industry Canada

This Class A digital apparatus meets all requirements of the Canadian Interference-Causing Equipment Regulations.

Cet appareil numérique de la classe A  
respecte toutes les exigences du  
Règlement Canadien sur le matériel brouilleur.

## FCC Part 68 Telecom

1. This equipment complies with part 68 of the Federal Communications Commission Rules. On the outside surface of this equipment is a label that contains, among other information, the FCC registration number. This information must be provided to the telephone company.

2. As indicated below, the suitable jack (Universal Service Order Code connecting arrangement) for this equipment is shown. If applicable, the facility interface codes (FIC) and service order codes (SOC) are shown.
3. An FCC compliant telephone cord and modular plug is provided with this equipment. This equipment is designed to be connected to the telephone network or premises wiring using a compatible modular jack that is Part 68 compliant. See installation instructions for details.
4. If this equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. If advance notice is not practical, the telephone company will notify the customer as soon as possible.
5. The telephone company may make changes in its facilities, equipment, operation, or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice to allow you to make necessary modifications to maintain uninterrupted service.
6. If trouble is experienced with this equipment (the model of which is indicated below), please contact Multi-Tech Systems, Inc. at the address shown below for details of how to have repairs made. If the equipment is causing harm to the network, the telephone company may request you to remove the equipment from the network until the problem is resolved.
7. No repairs are to be made by you. Repairs are to be made only by Multi-Tech Systems or its licensees. Unauthorized repairs void registration and warranty.
8. Manufacturer: Multi-Tech Systems, Inc.  
 Trade name: MultiVOIP  
 Model number: MVPFX2-2/4/8  
 FCC registration number: US: AU7DDNAN46050  
 Modular jack (USOC): RJ-48C  
 Service center in USA: Multi-Tech Systems, Inc.  
 2205 Woodale Drive  
 Mounds View, MN 55112  
 Tel: (763) 785-3500  
 FAX: (763) 785-9874

## Canadian Limitations Notice

**Notice:** The Industry Canada label identifies certified equipment. This certification means that the equipment meets certain telecommunications network protective, operational and safety requirements. The Department does not guarantee the equipment will operate to the user's satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be made by an authorized Canadian maintenance facility designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

**Caution:** Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.

## WEEE Statement

(Waste Electrical and Electronic Equipment)

July, 2005

The WEEE directive places an obligation on EU-based manufacturers, distributors, retailers and importers to take-back electronics products at the end of their useful life. A sister Directive, ROHS (Restriction of Hazardous Substances) complements the WEEE Directive by banning the presence of specific hazardous substances in the products at the design phase. The WEEE Directive covers all Multi-Tech products imported into the EU as of August 13, 2005. EU-based manufacturers, distributors, retailers and importers are obliged to finance the costs of recovery from municipal collection points, reuse, and recycling of specified percentages per the WEEE requirements.

### Instructions for Disposal of WEEE by Users in the European Union

The symbol shown below is on the product or on its packaging, which indicates that this product must not be disposed of with other waste. Instead, it is the user's responsibility to dispose of their waste equipment by handing it over to a designated collection point for the recycling of waste electrical and electronic equipment. The separate collection and recycling of your waste equipment at the time of disposal will help to conserve natural resources and ensure that it is recycled in a manner that protects human health and the environment. For more information about where you can drop off your waste equipment for recycling, please contact your local city office, your household waste disposal service or where you purchased the product.



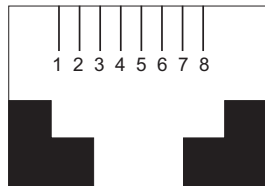
---

## **Appendix A: Cable Pinouts**

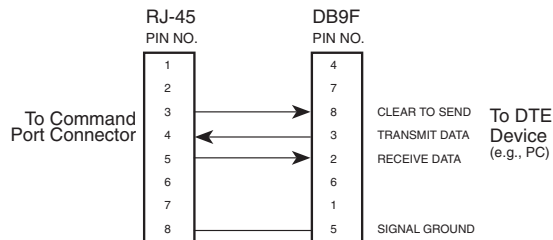
# Appendix A: Cable Pinouts

## Command Cable

### RJ-45 Connector



### End-to-End Pin Info



RJ-45 connector plugs into Command Port of MultiVOIP.  
 DB-9 connector plugs into serial port of command PC (which runs MultiVOIP configuration software).

## Ethernet Connector

The functions of the individual conductors of the MultiVOIP’s Ethernet port are shown on a pin-by-pin basis below.

RJ-45 Ethernet Connector	Pin	Circuit Signal Name
	1	TD+ Data Transmit Positive
	2	TD- Data Transmit Negative
	3	RD+ Data Receive Positive
	6	RD- Data Receive Negative

Pin Functions (FXS/FXO Interface)			
FXS Pin	Description	FXO Pin	Description
2	N/C	2	N/C
3	Ring	3	Tip
4	Tip	4	Ring
5	N/C	5	N/C

---

## **Appendix B: TCP/UDP Port Assignments**

## Well Known Port Numbers

The following description of port number assignments for Internet Protocol (IP) communication is taken from the Internet Assigned Numbers Authority (IANA) web site ([www.iana.org](http://www.iana.org)).

“The Well Known Ports are assigned by the IANA and on most systems can only be used by system (or root) processes or by programs executed by privileged users. Ports are used in the TCP [RFC793] to name the ends of logical connections which carry long term conversations. For the purpose of providing services to unknown callers, a service contact port is defined. This list specifies the port used by the server process as its contact port. The contact port is sometimes called the "well-known port". To the extent possible, these same port assignments are used with the UDP [RFC768]. The range for assigned ports managed by the IANA is 0-1023.”

Well-known port numbers especially pertinent to MultiVOIP operation are listed below.

## Port Number Assignment List

### Well-Known Port Numbers

Function	Port Number
tftp	69
SIP	5060
SysLog	514
http	80



# INDEX

- 1 LED ..... 11
- abbreviated dialing, inter-office
  - T1 ..... 102
- Accept Any Number (inbound)
  - T1 ..... 110
- Accept Any Number (outbound) field
  - T1 ..... 105
- accessing Call Progress (Statistics) screen ..... 133
- accessing configuration parameter groups ..... 62
- accessing interface parameters ..... 77
- accessing IP Parameters screen ..... 26, 63
- accessing IP Statistics screen ..... 138
- accessing logs screen ..... 92
- accessing Regional Parameters ..... 89
- accessing RTP Parameters screen ..... 98
- accessing System Information screen ..... 96
- accessing Voice/FAX Parameters screen ..... 68
- Add Prefix (inbound) field
  - T1 ..... 110
- Add Prefix (outbound) field
  - T1 ..... 106
- Advanced Features field group ..... 76
- airflow ..... 50
- allowing pop-ups with Web GUI ..... 62
- analog SIP-only voip product family ..... 6
- Append SIP Proxy Domain Name in User ID (Call Signaling) ..... 88
- Auto Disconnect field group ..... 75
- AutoCall field ..... 72
- Automatic Disconnection field ..... 75
- bandwidth, coder ..... 71
- battery caution ..... 45
- Boot LED ..... 11
  - MVPFX2-2 ..... 54
  - MVPFX2-4/8 ..... 53
- Boot Version
  - System Info ..... 96, 131
- booting time ..... 11
- box contents
  - verifying ..... 47
- built-in modem
  - setup in Regional Parameters screen ..... 61
- Bytes Received (call progress) field ..... 135
- Bytes Sent (call progress) field ..... 135
- cabling procedure
  - MVPFX2-2 ..... 53
  - MVPFX2-4 ..... 52
  - MVPFX2-8 ..... 52
- Cadence field ..... 91
- cadences, signaling ..... 89
- Call Control PHB field ..... 64
- Call Control Status
  - Call Progress Details (statistics) field ..... 137
- Call Duration field ..... 75
- Call Progress (Statistics) ..... 133
- Call Progress Details (statistics) field definitions 134, 135, 136, 137
- Call Progress Details (statistics) screen fields
  - Channel ..... 134
  - Duration ..... 134
  - Mode ..... 134
  - Voice Coder ..... 134
  - Packets Sent ..... 135
  - Packets Received ..... 135
  - Bytes Sent ..... 135
  - Bytes Received ..... 135
  - Packets Lost ..... 135
  - Outbound Digits ..... 135
  - Prefix Matched ..... 135
  - Call Status ..... 137
  - Silence Compression ..... 137
  - Disconnect ..... 137
  - Gateway Name (from and to) ..... 136
  - IP Address (from and to) ..... 136
  - Options (from) ..... 136
  - Gateway Name (from ..... 136
  - IP Address (from ..... 136
  - Options (from ..... 136
  - Gateway Name (to ..... 136
  - IP Address (to ..... 136
- Call Signaling screen fields
  - Append SIP Proxy ... ..... 88
  - Password ..... 89
  - Port Number ..... 88
  - Proxy Domain Name / IP Address ..... 88
  - Re-Registration Time ..... 89
  - Signaling Port ..... 88
  - Use SIP Proxy ..... 88
  - User Name ..... 89
- Call Status (call progress) field ..... 137
- Caller ID enable
  - FXO ..... 83
  - FXS Loop Start ..... 80, 81
- Caller ID examples ..... 86
- Caller ID fields
  - FXO ..... 83
  - FXS Loop Start ..... 81
- Caller ID Type
  - FXO ..... 83
  - FXS Loop Start ..... 80
- Canadian Class A requirements ..... 178
- Canadian Limitations Notice (regulatory) ..... 179
- CD, MultiVOIP ..... 13
- Channel (call progress) field ..... 134
- channel capacity ..... 7
- Channel Number (inbound) field
  - T1 ..... 110
- channel tracing on/off (logging) ..... 95

Clear (IP Statistics) button.....	139	debugging messages.....	94
coder		Default (Voice/FAX) field.....	69
bandwidth, max.....	71	delay, packets.....	73
G.711.....	71	Description (callee location)	
G.723.1.....	71	T1.....	110
G.726.....	71	Description (callee, outbound phonebook)	
G.729.....	71	T1.....	106
Coder field.....	71	Destination Pattern (outbound) field	
coder options		T1.....	106
packetization rates and.....	98	destination patterns, discussion	
Coder Parameters field group.....	71	T1.....	102
coder types (voice/fax, RTP packetization).....	99	Detection Range, Flash Detection Range fields	
COL LED.....	11	FXS Loop Start.....	80
command cable pinout.....	182	Dialing Options (FXO) fields.....	85
command modem		DiffServ and IP datagram.....	66
and Regional Parameters screen.....	61	DiffServ PHB (Per Hop Behavior) value.....	64
Command Modem		dimensions.....	12
setup for.....	61	Disconnect (call progress) field.....	137
Command PC		disconnection criteria, FXO.....	84, 85
COM port requirement.....	12	DNS Server IP Address (IP Parameters) field.....	67
non-dedicated use of.....	12	downloading firmware, machine perspective.....	149
operating system.....	12	downloads vs. uploads (FTP).....	149
compatibility, H.450 with H.323, not with SIP.....	8	DTMF	
compression, silence.....	76	extended.....	85
computer requirements.....	12	standard.....	85
configuration of voip		DTMF frequency chart.....	85
local versus remote.....	58	DTMF In/Out of Band field.....	70
Configuration Parameter Groups, accessing.....	62	DTMF inbound.....	70
configuration procedure		DTMF out of band.....	70
detailed.....	61	DTMF Tone (FXO disconnect criteria) field.....	84
configuration procedure, local		Duration (call progress) field.....	134
summary.....	61	Duration (DTMF) field.....	70
Configuration Version		Dynamic Jitter Buffer field.....	73
System Info.....	97	Dynamic Jitter field group.....	73
Configur-ation Version		Dynamic Jitter fields.....	74
System Information.....	131	Echo Cancellation field.....	76
configuration, local.....	59	echo, removing.....	76
configuration, phonebook		Edit Entry screen	
T1.....	102	inbound.....	112
configuration, saving.....	100	outbound.....	111
Configuring MultiVOIP phonebooks, general		EMC, Safety, R&TTE Directive Compliance.....	178
T1.....	102	Enable Console Messages field.....	94
Consecutive Packets Lost field.....	75	Enable DHCP (IP Parameters) field.....	65
Console Message Settings, Filters for.....	95	Enable DNS (IP Parameters) field.....	67
console messages, enabling.....	93	enabling web browser GUI.....	62
console parameters tracked.....	95	ethernet cable pinout.....	182
contacting technical support.....	176	Ethernet interface.....	7
coordinated phonebook entries		European Community Directives.....	178
T1.....	102	factory repair for customers U.S. & Canada.....	175
Copy Channel command (Interface Parameters).....	78	FAQ for MultiVOIPs.....	6
Copy Channel command (Voice/Fax Parameters).....	69	fax machine	
Copy Channel field.....	69	connecting to voip (MVPFX2-2).....	54
Country/Region (tone schemes) field.....	90	connecting to voip (MVPFX2-4/8).....	52
Current Loss (FXO disconnect criteria) field.....	84	FCC Declaration.....	178
Current Loss field		FCC Part 68 Telecom rules.....	178
FXS Loop Start.....	80	FCC registration number.....	179
Current Loss Timer (FXO disconnect criteria) field.....	84	FCC rules, Part 15.....	178
data capacity.....	7	FDX LED.....	11
data compression.....	8	file system image file	

- Java cache clearing, and.....157, 169
- Filters (Console Message Settings).....95
- Filters button (Console Message Settings) .....94
- Firmware Version
  - System Information .....131
- Firmware Version (System Info) .....96
- firmware version, identifying .....146
- firmware, obtaining updated .....147
- Flash Generation field
  - FXO .....84
- forgotten password.....143
- Frame Type field.....64
- free calls
  - T1 .....102
- frequencies, touch tone .....85
- Frequency 1 (tone pair scheme).....91
- Frequency 2 (tone pair scheme).....91
- frequency, power .....12
- front panel.....11
- FTP client program .....149
- FTP client program, obtaining .....151
- FTP client programs
  - graphic vs. textual orientation.....156
- FTP file transfers
  - using FTP client program .....151
  - using web browser .....151
- FTP Server Enable (IP Parameters) field.....67
- FTP Server function
  - as added feature .....149
  - enabling .....151
- FTP Server, contacting .....152
- FTP Server, invoking download/transfer
  - using FTP client program .....155
  - using web browser .....153
- FTP Server, logging in.....152
- FTP transfers
  - file types .....149, 152
  - server location.....149
- function tracing on/off (logging) .....95
- FXO disconnection criteria .....84
- FXO disconnection, triggering of .....85
- FXO Interface parameter definitions .....83
- FXO Interface Parameter definitions .....83, 84, 85
- FXO Interface Parameter fields
  - Caller ID Enable .....83
  - Caller ID Type .....83
  - Current Loss .....84
  - Current Loss Detect Timer .....84
  - Flash Generation .....84
  - Inter Digit Regeneration Timer.....84
  - Inter Digit Timer (dialing) .....85
  - No Response Timer .....84
  - Regeneration (dialing) .....85
  - Ring Count.....84
  - Tone Detection.....84
- FXO Parameter fields
  - Interface Type.....83
  - Select Channel .....83
- FXO Parameters.....82
- FXS Loop Start
  - Interface Type.....79
- FXS Loop Start Interface parameter definitions .....79
- FXS Loop Start Interface Parameter fields
  - Caller ID enable.....81
  - Caller ID Enable .....80
  - Caller ID Type .....80
  - Current Loss .....80
  - Flash Detection Range.....80
  - Inter Digit Regeneration Timer.....81
  - Inter Digit Timer.....81
  - Regeneration.....81
  - Ring Count.....80
- FXS Loop Start Parameter fields
  - Select Channel .....79
- FXS Loop Start Parameters .....79
- FXS/FXO connector
  - MVPFX2-2 .....54
  - MVPFX2-4/8.....52
- G711 coders (RTP packetization, voice/fax).....99
- G723 coders (RTP packetization, voice/fax).....99
- G726 coders (RTP packetization, voice/fax).....99
- G727 coders (RTP packetization, voice/fax).....99
- G729 coders (RTP packetization, voice/fax).....99
- Gain 1 (tone pair scheme).....91
- Gain 2 (tone pair scheme).....91
- Gateway (IP Parameters) field.....65
- Gateway Name (IP Parameters) field .....65
- grounding
  - in rack installations .....50
  - MVPFX2-2 .....54
- H.450 features, incompatible with SIP .....8
- IANA .....184
- identifying current firmware version .....146
- in band, DTMF .....70
- Inbound Phone Book Add Entry screen
  - T1 .....110
- Inbound Phone Book Add Entry screen field definitions
  - T1 .....110
- Inbound Phone Book Add Entry screen fields (T1)
  - Accept Any Number .....110
  - Add Prefix .....110
  - Channel Number.....110
  - Description (callee location).....110
  - Registration Option Parameters .....110
  - Remove Prefix .....110
- Inbound Phone Book Edit Entry screen.....112
  - T1 .....111
- Inbound Phonebook entries, list
  - T1 .....108
- inbound vs. outbound phonebooks
  - T1 .....102
- Industry Canada requirements .....178
- info sources
  - IP details .....59
  - telephony interface details .....60
- Initial Jitter Value field.....74
- Input Gain field.....69
- installation

- airflow.....50
- in a nutshell.....13
- in rack .....49
- software (detailed) .....56
- installation prerequisites .....59, 60
- installation, mechanical .....7
- Inter Digit Regeneration Time
  - FXO .....85
- Inter Digit Regeneration Timer
  - FXS Loop Start .....81
- Inter Digit Timer (dialing) field
  - FXO .....85
  - FXS Loop Start .....81
- interface parameters, accessing.....77
- interface parameters, setting .....77
- Interface Type
  - FXO .....83
  - FXS Loop Start .....79
- inter-office dialing
  - T1.....102
- inter-operation (analog)
  - with T1/E1 voips.....7
- inter-operation with phone system.....8
- IP Address (IP Parameters) field.....65
- IP Address (IP Statistics) field.....139
- IP Address (outbound phonebook)
  - T1.....106
- IP address, SysLog Server .....94
- IP datagram and DiffServ .....66
- IP Mask field .....65
- IP parameter definitions.....64, 65, 67
- IP Parameter fields
  - Frame Type.....64
- IP Parameter screen fields
  - Enable DNS .....67
- IP Parameters screen fields
  - Call Control PHB.....64
  - DiffServ .....64
  - DNS Server IP Address .....67
  - Enable DHCP.....65
  - Enable SRV .....67
  - FTP Server Enable.....67
  - Gateway .....65
  - Gateway Name.....65
  - IP Address.....65
  - IP Mask.....65
  - Voip Media PHB .....64
- IP Parameters screen, accessing.....26, 63
- IP Statistics field
  - IP Address.....139
- IP Statistics field definitions .....139, 140
- IP Statistics fields
  - Clear.....139
  - Received (RTCP Packets).....141
  - Received (RTP Packets) .....141
  - Received (TCP Packets) .....140
  - Received (Total Packets) .....139
  - Received (UDP Packets).....140
  - Received with errors (RTCP Packets) .....141
  - Received with errors (RTP Packets).....141
  - Received with errors (TCP Packets).....140
  - Received with errors (Total Packets).....140
  - Received with errors (UDP Packets) .....140
  - Transmitted (RTCP Packets) .....141
  - Transmitted (RTP Packets).....141
  - Transmitted (TCP Packets).....140
  - Transmitted (Total Packets).....139
  - Transmitted (UDP Packets) .....140
- IP Statistics function.....138
- Java cache clearing
  - file system image file updating, and .....157, 169
  - mvpfs-img file updating, and.....157, 169
- Java software
  - installing .....56
- jitter buffer.....73
- Jitter Value field .....75
- jitter, dynamic.....73
- Kernel Version
  - System Information .....97, 131
- Knowledge Base (online, for MultiVOIPs) .....6
- LED definitions
  - 1, 2, 3, ...8 .....11
  - Boot .....11
  - COL .....11
  - Ethernet.....11
  - FDX .....11
  - LNK.....11
  - Power.....11
  - SPD.....11
  - Voice/Fax 1 .....11
  - Voice/Fax 2 .....11
- LED indicators
  - channel operation.....10
  - general operation .....10
- LED indicators, active .....10
- LED types .....10
- lifting
  - precaution about .....45
- limitations notice (regulatory), Canadian .....179
- limited warranty.....175
- lithium battery caution.....45
- LNK LED .....11
- loading of weight in rack .....50
- local configuration.....59
- local configuration procedure
  - summary .....61
- local voip configuration.....58
- log reporting method, setting.....92
- logging options .....93
- logging update interval .....93
- Logs screen definitions.....93
- Logs screen field definitions.....94
- Logs screen parameters
  - Enable Console Messages.....94
  - Filters.....94
  - IP Address (SysLog Server) .....94
  - Online Statistics Update Interval .....94
  - Port (SysLog Server) .....94

- SysLog Server Enable.....94
- logs screen, accessing.....92
- long-distance call savings
  - T1.....102
- lost packets, consecutive.....75
- lost password .....143
- Mac Address
  - System Info.....97, 131
- mains frequency.....12
- Max bandwidth (coder).....71
- Maximum Jitter Value field.....74
- Minimum Jitter Value field.....73
- Mode (call progress) field.....134
- modem, command
  - and Regional Parameters Country Selection.....61
- modem, remote configuration/command
  - setup for .....61
- mounting.....7
- mounting in rack.....49
  - procedure for.....51
  - safety.....45, 50
- MSP Version
  - System Information .....97, 131
- MultiVOIP FAQ (on MTS web site) .....6
- MultiVOIP general operation functions, option
  - descriptions .....142
- MultiVOIP software
  - moving around in.....62
- mvpfs-img file update
  - Java cache clearing, and.....157, 169
- MVPFX2-2
  - cabling procedure.....53
  - grounding.....54
  - unpacking.....48
- MVPFX2-4
  - cabling procedure.....52
  - unpacking.....47
- MVPFX2-8
  - cabling procedure.....52
  - unpacking.....47
- Netcoder coders (RTP packetization, voice/fax) .....99
- Network Disconnection field .....75
- No Response Timer
  - FXO .....84
- obtaining updated firmware .....147
- Online Statistics Update Interval field (Logs) .....94
- operating temperature .....50
- out of band, DTMF.....70
- Outbound Digits Sent (call progress) field .....135
- Outbound Phone Book Add Entry field definitions
  - T1.....106, 107
- Outbound Phone Book Add Entry fields (T1)
  - Accept Any Number .....105
  - Add Prefix.....106
  - Advanced button.....107
  - Description.....106
  - Destination Pattern.....106
  - IP Address.....106
  - Remove Prefix .....106
- SIP Port Number.....107
- SIP URL .....107
- Total Digits.....106
- Transport Protocol (SIP).....107
- Use Proxy (SIP).....107
- Outbound Phone Book Edit Entry screen .....111
  - T1.....111
- Outbound Phonebook Add Entry field definitions
  - T1.....105
- Outbound Phonebook Add Entry screen
  - T1.....104
- Outbound Phonebook entries, list
  - T1.....103
- outbound vs. inbound phonebooks
  - T1.....102
- Output Gain field .....69
- packet priority and DiffServ .....66
- packetization (RTP), ranges & increments .....99
- packetization rates
  - coder options and.....98
- Packets Lost (call progress) field.....135
- Packets Received (call progress) field .....135
- Packets Sent (call progress) field.....135
- packets, consecutive lost.....75
- parameters tracked by console.....95
- Password (Call Signaling) field .....89
- password, lost/forgotten.....143
- password, setting
  - web browser GUI.....143
- patents .....2
- PBX characteristics, variations in
  - T1.....127
- PBX interaction .....8
- personnel requirement
  - for rack installation.....50
  - to lift during installation .....51
  - to lift unit during installation .....45
- Phone Book Version
  - System Info.....97
  - System Information .....131
- Phone Number (Voice/FAX – AutoCall) field .....72
- Phone Signaling Tones & Cadences.....89
- phone/IP details
  - importance of writing down.....59
- phonebook configuration.....58
- Phonebook Configuration Procedure
  - T1.....102
- Phonebook Configuration screen
  - T1.....102
- phonebook entries, coordinating
  - T1.....102
- phonebook, objectives & considerations
  - T1.....102
- phonebooks, inbound vs. outbound
  - T1.....102
- pinout
  - command cable .....182
  - ethernet cable.....182
- pop-ups

allowing with Web GUI.....	62	RFC2474.....	64
Port field, SysLog Server.....	94	RFC2597.....	64
Port Number (Call Signaling) field.....	88	RFC3246.....	64
power consumption.....	12	RFC768.....	184
power frequency.....	12	RFC793.....	184
Power LED.....	11	Ring Count	
Prefix Matched (call progress) field.....	135	FXO.....	84
prerequisites		Ring Count field	
for technical configuration.....	59	FXS Loop Start.....	80
product CD.....	13	RTP packetization, ranges & increments.....	99
use in software installation.....	56	RTP Parameters screen.....	99
Proxy Domain Name / IP Address field.....	88	Safety Recommendations for Rack Installations.....	50
Proxy Parameters.....	88	safety warnings.....	45
quality-of-service.....	8	<b>Safety Warnings Telecom</b> .....	46
rack mounting		Save & Apply command.....	100
grounding.....	50	saving configuration.....	100
safety.....	45, 50	Saving the MultiVOIP Configuration.....	100
rack mounting instructions.....	49	savings on toll calls	
rack mounting procedure.....	51	T1.....	102
rack, equipment		Select Channel	
weight capacity of.....	50	FXO.....	83
rack-mountable voip models.....	45	FXS Loop Start.....	79
Received (RTCP Packets, IP Stats) field.....	141	Select Channel field.....	69
Received (RTP Packets, IP Stats) field.....	141	Selected Coder field.....	71
Received (TCP Packets, IP Stats) field.....	140	Sequence (FXO) field.....	85
Received (Total Packets, IP Stats) field.....	139	Set Log Reporting Method.....	92
Received (UDP Packets, IP Stats) field.....	140	Set Password (web browser GUI) , command.....	143
Received with Errors (RTCP Packets, IP Stats) field.....	141	Set Regional Parameters.....	89
Received with Errors (RTP Packets, IP Stats) field.....	141	Set Telephony Interface Parameters.....	77
Received with Errors (TCP Packets, IP Stats) field.....	140	Set Voice/FAX Parameters.....	68
Received with Errors (Total Packets, IP Stats) field.....	140	setting IP parameters.....	26, 63
Received with Errors (UDP Packets, IP Stats) field.....	140	setting password	
Regeneration (dialing, FXO) field.....	85	web browser GUI.....	143
Regeneration field		setting RTP Parameters.....	99
FXS Loop Start.....	81	setup, saving.....	100
Regional Parameter definitions.....	90, 91	signaling cadences.....	89
Regional Parameter fields		Signaling Port (Call Signaling) field.....	88
Cadence.....	91	signaling tones.....	89
Country/Region (tone schemes).....	90	signaling types	
Frequency 1.....	91	telephony interfaces (MVPFX2-2).....	54
Frequency 2.....	91	telephony interfaces (MVPFX2-4/8).....	52
Gain 1.....	91	Silence Compression (call progress) field.....	137
Gain 2.....	91	Silence Compression field.....	76
type (of tone).....	91	Silence Detection (FXO disconnect criteria) field.....	85
regional parameters, setting.....	89	Silence Detection Type (FXO) field.....	85
Registerwith SIP Proxy (Inbound Phone Book)		Silence Timer (FXO disconnect criteria) field.....	85
T1.....	110	SIP Call Signaling Parameter definitions.....	88
Remote Configuration/Command Modem		SIP Fields (Outbound Phonebook)	
setup for.....	61	T1.....	107
remote voip configuration.....	58	SIP incompatibility with H.450 Supplementary Services.....	8
Remove Prefix (inbound) field		SIP Parameters.....	88
T1.....	110	SIP Port Number field	
Remove Prefix (outbound) field		T1.....	107
T1.....	106	SIP port number, standard	
repair procedures for customers U.S. & Canada.....	175	T1.....	107
Re-Registration Time (Call Signaling).....	89	SIP URL field	
Resolutions (MultiVOIP troubleshooting).....	6	T1.....	107
RFC 2833.....	70	software configuration	
RFC 3087.....	107	summary.....	56

- software installation
  - detailed.....56
- software loading.....56
- software, MultiVOIP
  - moving around in.....62
- software, MultiVOIP
  - screen-surfing in .....62
- sound quality, improving .....76
- SPD LED .....11
- supervisory signaling .....77
- supervisory signaling types
  - MVPFX2-2 .....54
  - MVPFX2-4/8 .....52
- Supplementary Services, incompatible with SIP .....8
- support, technical .....176
- SysLog client .....9
- SysLog client programs
  - availability .....171
  - features & presentation types .....173
- SysLog functionality.....9
- SysLog server .....9
- SysLog Server Enable field .....94
- SysLog Server function
  - as added feature .....171
  - capabilities of.....173
  - enabling .....172
  - location of.....171
- SysLog Server IP Address field.....94
- SysLog Server, enabling.....93
- System Information Parameters
  - Boot Version.....131
  - Configuration Version .....131
  - Kernel Version.....97, 131
  - Mac Address .....97, 131
  - MSP Version.....97, 131
  - Phone Book Version .....131
  - Up Time .....97, 131
- System Information screen
  - for op & maint .....130
- System Information screen, accessing .....96
- System Information update interval, setting .....96
  - for op & maint .....132
- table-top voip models .....45
- TCP/UDP compared
  - IP Statistics context.....139
  - T1.....107
- technical configuration
  - prerequisites to.....59
  - summary .....58
- technical configuration procedure
  - detailed.....61
  - summary .....61
- technical support.....176
- telecom safety warnings** .....46
- telephony interface parameters .....60
- telephony interface parameters, setting.....77
- telephony interfaces
  - uses of.....52, 54
- telephony signaling cadences.....89
- telephony signaling tones .....89
- temperature
  - operating.....50
- timeout interval
  - voips under SIP proxy server.....89
- toll-call savings
  - T1.....102
- tones, signaling .....89
- Total Digits (outbound) field
  - T1.....106
- touch tone frequencies .....85
- trace on/off (logging).....95
- Transmitted (RTCP Packets, IP Stats) field.....141
- Transmitted (RTP Packets, IP Stats) field .....141
- Transmitted (TCP Packets, IP Stats) field .....140
- Transmitted (Total Packets, IP Stats) field .....139
- Transmitted (UDP Packets, IP Stats) field.....140
- Transport Protocol (SIP) field
  - T1.....107
- Troubleshooting Resolutions for MultiVOIPs.....6
- Type (of tone, Regional Parameters) field.....91
- Type-of-Service IP header field & DiffServ.....66
- UDP/TCP compared
  - IP Statistics context .....139
  - T1.....107
- unpacking .....47
  - MVPFX2-2 .....48
  - MVPFX2-4 .....47
  - MVPFX2-8 .....47
- Up Time
  - System Info.....97, 131
- update interval (logging) .....93
- updated firmware, obtaining .....147
- uploads vs. downloads (FTP) .....149
- Use Proxy (SIP) field
  - T1.....107
- Use SIP Proxy field .....88
- User Name (Call Signaling) field .....89
- variations in PBX characteristics
  - T1.....127
- version, firmware.....146
- Voice Coder (call progress) field.....134
- voice delay.....73
- Voice Gain field .....69
- voice packets, consecutive lost .....75
- voice packets, delayed .....73
- voice quality, improving.....76
- Voice/Fax 1 LED.....11
- Voice/Fax 2 LED.....11
- Voice/FAX Parameter AutoCall fields
  - Auto Call .....72
  - Phone Number .....72
- Voice/FAX Parameter definitions .....74, 75
- Voice/FAX Parameter Definitions .....69, 70, 71, 73, 76
- Voice/FAX Parameter fields
  - AutoCall .....72
  - AutoCall fields.....72
  - Out-of-Band Mode (DTMF).....70
- Voice/FAX Parameter fields

Copy Channel .....	69	Voice/FAX Parameter fields	
Default .....	69	Echo Cancellation.....	76
DTMF In/Out of Band .....	70	Voice/FAX Parameters screen, accessing .....	68
Duration (DTMF) .....	70	Voice/FAX parameters, setting .....	68
Input Gain .....	69	Voip Caller ID Case #1 –telco standard CID enters voip system.....	86
Output Gain .....	69	Voip Caller ID Case #4 – Remote FXS call on H.323 voip system.....	86
Select Channel .....	69	Voip Media PHB field.....	64
Voice Gain.....	69	voip software	
Voice/FAX Parameter fields		host PC.....	12
Dynamic Jitter Buffer .....	73	voip system example, digital & analog, with phonebook details	
Voice/FAX Parameter fields		T1.....	119
Minimum Jitter Value.....	73	voip system example, digital only, with phonebook details	
Voice/FAX Parameter fields		T1.....	113
Maximum Jitter Value .....	74	warnings, safety .....	45
Voice/FAX Parameter fields		warranty.....	175
Initial Jitter Value .....	74	web browser GUI, enabling.....	62
Voice/FAX Parameter fields		weight .....	12
Automatic Disconnection .....	75	weight loading	
Voice/FAX Parameter fields		in rack .....	50
Jitter Value.....	75	weight of unit	
Voice/FAX Parameter fields		lifting precaution.....	45
Call Duration .....	75	personnel requirement .....	45
Voice/FAX Parameter fields		Well Known Ports .....	184
Consecutive Packets Lost .....	75	well-known port, SIP	
Voice/FAX Parameter fields		T1.....	107
Network Disconnection .....	75		
Voice/FAX Parameter fields			
Silence Compression .....	76		





S000411A